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Rozenblit

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[54] **COMMUNICATION NETWORK WITH HIDDEN CALLING NUMBER CAPABILITY**

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[21] Appl. No.: **757,560**

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[51] **Int. Cl.**[6] **H04M 1/57**; H04M 3/42

[52] **U.S. Cl.** **379/246**; 379/142; 379/201

[58] **Field of Search** 379/201, 142, 379/127, 196, 197, 198, 222, 223, 215, 245, 246, 247

[56] **References Cited**

U.S. PATENT DOCUMENTS

4,162,377	7/1979	Mearns	379/127
4,277,649	7/1981	Sheinbein	379/201
5,008,930	4/1991	Gawrys et al.	379/211
5,033,076	7/1991	Jones et al.	379/142
5,278,894	1/1994	Shaw	379/67
5,289,542	2/1994	Kessler	379/142
5,341,411	8/1994	Hashimoto	379/142
5,341,414	8/1994	Popke	379/142
5,475,744	12/1995	Ikeda	379/142
5,497,414	3/1996	Bartholomew	379/142
5,521,969	5/1996	Paulus et al.	379/201
5,526,406	6/1996	Luneau	379/142
5,530,741	6/1996	Rubin	379/142
5,548,636	8/1996	Bannister et al.	379/201
5,590,184	12/1996	London	379/142
5,604,792	2/1997	Solomon et al.	379/142
5,636,209	6/1997	Perlman	370/281
5,666,405	9/1997	Weber	379/142
5,668,852	9/1997	Holmes	379/142
5,668,862	9/1997	Bannister et al.	379/201

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[57] **ABSTRACT**

A method and apparatus are disclosed for delivering calling line information in a manner which preserves the privacy interests of the caller while providing useful information to the called party. The caller initiates a call from a caller terminal of a communication network. A destination central office or other network element receives the call along with a complete calling line number and delivers partial calling line information to the called party prior to connecting the call. The partial calling line information may include an NPA-NXX portion of an NPA-NXX-XXXX calling line number, geographical information such as state, county, city, township and zip code associated with the calling line, and/or an indication as to whether the call is from a residence, business, pay phone, cellular phone or other particular source such as a hospital or hotel. The central office or other network element may store the full calling line number in an incoming call log maintained for the called party, and a name selected by the called party may be associated with the calling line number. The central office will then deliver the selected name as part of the partial calling information when subsequent calls are placed from the calling line to the called party. This allows the called party to perform selective screening, call transfer and other functions without ever learning the full calling line number. The called party can also direct the central office to utilize the full calling line number stored therein to initiate a return call to the corresponding calling line.

15 Claims, 2 Drawing Sheets

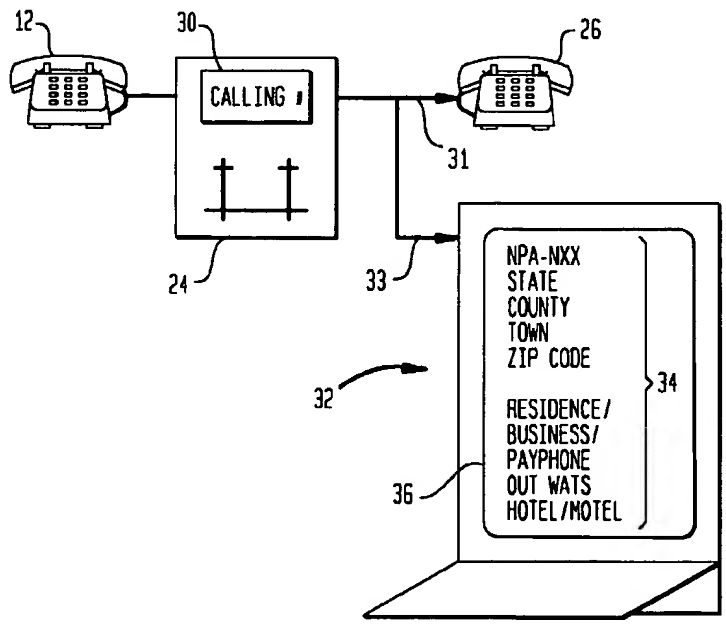


FIG. 1

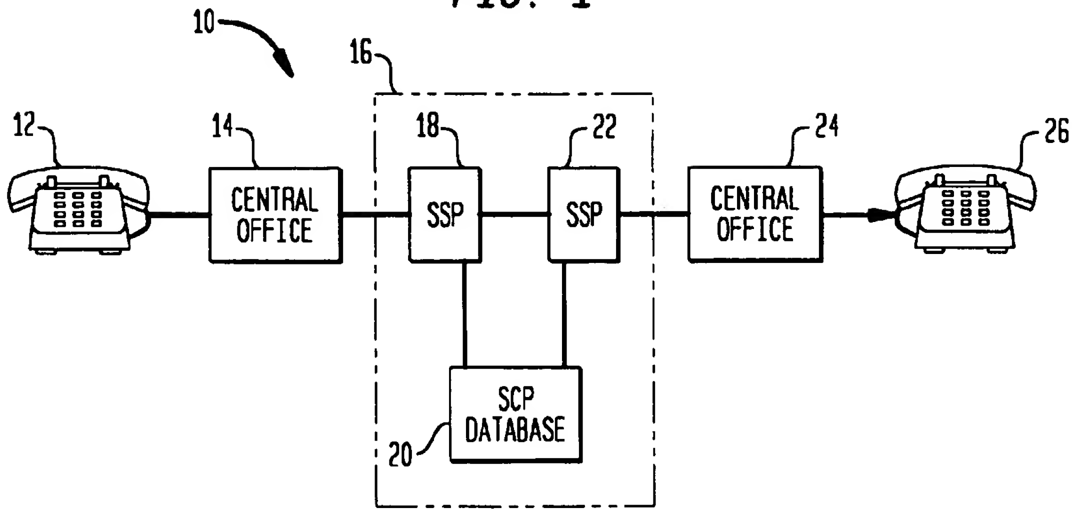

10
16
12
14
CENTRAL OFFICE
18
SSP
22
SSP
24
CENTRAL OFFICE
26
SCP DATABASE
20

FIG. 2

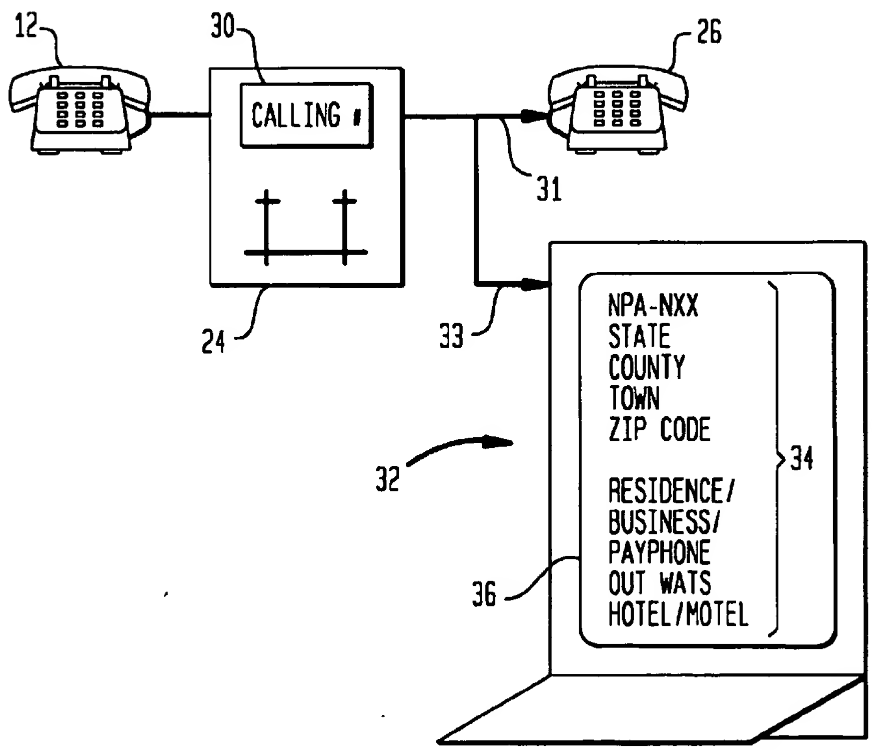
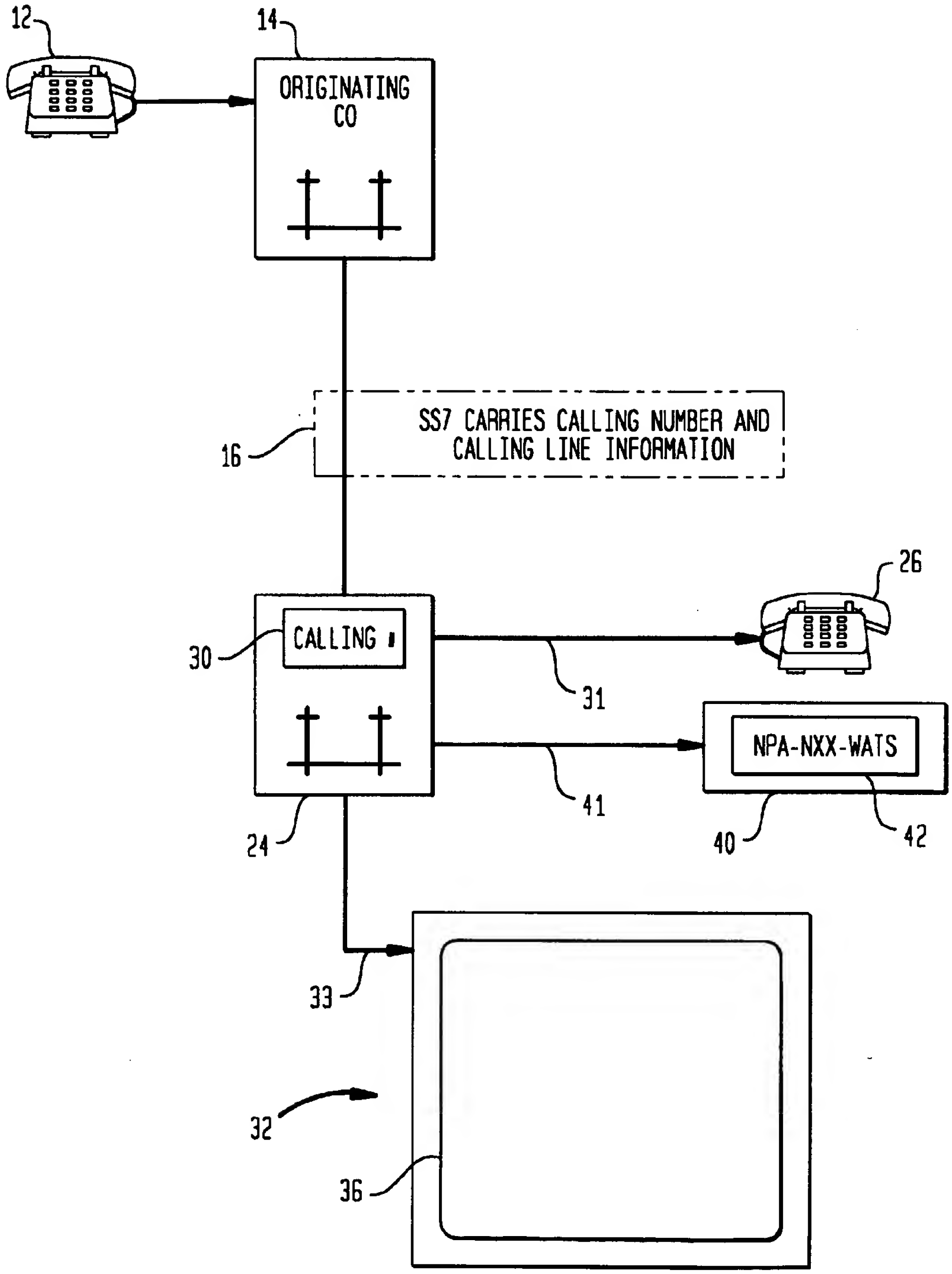

12
30
CALLING
26
31
24
33
NPA-NXX
STATE
COUNTY
TOWN
ZIP CODE
RESIDENCE/
BUSINESS/
PAYPHONE
OUT WATS
HOTEL/MOTEL
34
32
36

FIG. 3


12
14
ORIGINATING CO
16
SS7 CARRIES CALLING NUMBER AND CALLING LINE INFORMATION
26
30
CALLING #
31
NPA-NXX-WATS
41
24
40
42
33
32
36

COMMUNICATION NETWORK WITH HIDDEN CALLING NUMBER CAPABILITY

FIELD OF THE INVENTION

The present invention relates generally to communication networks and more particularly to communication networks in which calling line identification information is delivered to a called party.

BACKGROUND OF THE INVENTION

Many telephone networks are configured to provide a calling number delivery service in which a calling line number is displayed to a called party prior to connection of a call. Such services are also commonly referred to as caller identification or caller ID services. The called party uses the displayed calling line number to determine whether or not to accept the call. The called party may also arrange in advance to block or to transfer incoming calls from certain calling line numbers, or to log calls for later return. The called parties are generally charged for this calling number delivery service, and the service therefore represents a substantial source of revenue for network service providers.

Existing caller ID services typically involve the delivery of the entire calling line number to the called party. This presents a significant problem in that it may infringe on the privacy rights of the caller. For example, an abused spouse would be unable to call home from a shelter without the possibility of revealing the exact location of the shelter to the abuser via the displayed calling line number. The privacy problem has been addressed by allowing calling parties the option to block the display of their calling line number to called parties. For example, a particular caller can simply instruct a network provider not to deliver their calling line number to any called party. However, it is expected that the size of the caller ID customer market would be likely to decrease as a result of the availability of caller ID blocking. The provision of caller ID blocking therefore does not resolve the privacy problems of calling line number delivery in a manner which maximizes both the benefit of the service to customers, and the revenue available to service providers.

As is apparent from the above, a need exists for an improved calling line information delivery technique which preserves the privacy rights of the caller without unduly restricting the flow of useful information to the called party.

SUMMARY OF THE INVENTION

The present invention provides a method and apparatus for delivery of calling line information to a called party. The information is delivered such that the full calling line number is hidden from the called party, while other useful calling line information is made available to the called party. This partial calling line information may include, for example, calling line area code, state, county, town and zip code, and an indication as to whether the call is from a business, residence, pay phone, cellular phone or a specific location such as a hospital or hotel. Such information assists the called party in determining how to handle the call, while preserving the privacy rights of the caller.

An exemplary embodiment of the invention operates as follows. A caller initiates a call from a caller terminal of a communication network. A destination central office or other network element receives the call along with a complete calling line number and class of service information, and delivers partial calling line information to the called party prior to connecting the call. The partial calling line information may include an NPA-NXX portion of an NPA-NXX-XXXX calling line number along with at least a portion of the above-noted class of service information. The delivery of partial calling line information may therefore involve replacing the XXXX portion of the calling line number with a multi-character code indicative of the type of calling line. For example, the central office may deliver the number NPA-NXX-BUSN to indicate that the call is from a business, or NPA-NXX-RESD to indicate that the call is from a residence.

The central office or other network element may store the a given calling line number in an incoming call log maintained for the called party, and the called party may select a name to be stored in the central office along with the given calling line number. The central office may then deliver the selected name as part of the partial calling information when subsequent calls are placed from the given calling line to the called party. This allows the called party to perform selective screening, call transfer and other functions without ever learning the full calling line number. The called party can also direct the central office to utilize the full calling line number stored therein to initiate a return call to the corresponding calling line.

The present invention allows useful calling line information to be delivered to a called party while preserving the privacy rights of the calling party. The invention may be implemented with relatively simple software modifications to existing communication networks. Conventional customer premises equipment such as telephones and personal computers may be used. It should be noted that the present invention may be utilized in conjunction with rather than as a replacement for a conventional caller ID service. That is, a system configured in accordance with the invention could deliver a full calling line number as well as other partial calling line information for callers that have not requested privacy, while delivering only the partial calling line information for callers that have requested privacy. These and other advantages and features of the present invention will become more apparent from the accompanying drawings and the following detailed description.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram of an exemplary communication network in which the present invention may be implemented.

FIGS. 2 and 3 illustrate a calling line information delivery service in accordance with an exemplary embodiment of the invention.

DETAILED DESCRIPTION OF THE INVENTION

The present invention will be illustrated below in conjunction with an exemplary telephone network. It should be understood, however, that the invention is not limited to use with any particular type of communication network, but is instead more generally applicable to any calling line information delivery service. For example, the invention may be utilized for communication services provided over cable networks, satellite networks, computer networks such as the Internet as well as other types of communication networks. The term "call" is therefore intended to include not only telephone calls but more generally any communication initiated by or received in a caller terminal. A caller or called party "terminal" as used herein may be a telephone, a computer, a television set top box or any other device capable of initiating or receiving a call.

FIG. 1 shows an exemplary communication network **10** in which the calling line information delivery service of the present invention may be implemented. The exemplary communication network **10** is a telephone network providing voice communications between caller terminals. A caller initiates a call from a caller terminal **12**, which may be a telephone with a corresponding Plain Old Telephone Service (POTS) number associated therewith. The call initiated at caller terminal **12** is routed to an originating central office **14**, which may be a local exchange switch within a local exchange carrier (LEC) network. The central office **14** may include an AT&T 5ESS® switching system or other conventional switching system. The originating central office **14** directs the call to a switching network **16**, which may represent a portion of a long distance telephone network, an inter-exchange carrier (IXC) network, or an LEC network.

The caller terminal **12** typically has a unique calling line number associated therewith which may be used to identify the individual, household or business using the caller terminal **12**. For example, a POTS caller terminal typically has a calling line number of the form NPA-NXX-XXXX, in which NPA is the area code, NXX identifies the local exchange, and XXXX specifies the particular number. An Outward Wide Area Telephone Service (OUTWATS) caller terminal has a calling line number of the form 800-XXX-XXXX. The calling line number is also commonly referred to as a calling line identifier (CLI). Additional detail on conventional CLI-based call processing may be found in, for example, U.S. Pat. No. 4,277,649, which is incorporated by reference herein. Alternatives to the calling line number for identifying a particular caller include telephone credit card numbers.

The switching network **16** in this embodiment includes a first service switching point (SSP) **18**, a service control point (SCP) database **20**, and a second SSP **22**. Each of the SSPs **18, 22** may be a toll office with Common Channel Signalling (CCS) capability, such as a conventional AT&T 4ESS® switching system. The CCS capability provides a high-speed packet-switched data link which can be used to carry network control information to or from SCP database **20**. Additional details regarding network control using CCS may be found in, for example, U.S. Pat. Nos. 4,162,377 and 4,277,649, both of which are incorporated by reference herein. An exemplary type of CCS suitable for use in the present invention is CCS No. 7, also known as Signalling System 7 (SS7). Alternatively, each of the SSPs **18, 22** may be interexchange switches in a long distance carrier or IXC network, or local exchange switches in an LEC network. Calls placed at caller terminal **12** are directed by central office **14** to first SSP **18**. The first SSP **18** may access the SCP database **20** to obtain routing instructions which direct SSP **18** to route the call in a particular manner. The call is routed by SSP **18** to SSP **22**, which in turn routes the call to a destination central office **24**. The destination central office **26** connects the call to a called party terminal **26**.

An exemplary embodiment of a calling line information delivery service in accordance with the invention may include one or more of the following features: (1) partial yet useful calling line information is provided to the called party in a manner which preserves the anonymity of the caller; (2) an incoming call log may be maintained such that the called party is able to return a particular call at any time, from any caller terminal, without being made aware of the full calling line number; (3) the called party can assign a user-selected name to be associated with the calling line number of a particular incoming call without being made aware of the full calling line number; and (4) multiple incoming call logs can be created in user-selected formats such that the called party can block any further calls from any given caller without being made aware of the full calling line number. These and other aspects of the invention will be described in greater detail below. It should be noted that each of the above-noted features may be provided independently of the others in a given communication system. The particular feature or combination of features provided in a given system may vary depending upon the needs of service providers and customers.

FIG. 2 illustrates the delivery of partial calling line information in the exemplary communication network of FIG. 1. As noted above, a POTS telephone number generally is of the form NPA-NXX-XXXX where NPA represents the area code, NXX identifies the local exchange, and XXXX represents the remaining four digits of a particular telephone number. A caller initiates a call from caller terminal **12** having the NPA-NXX-XXXX calling line number. The call is routed in the manner previously described to the destination central office **24** which stores the complete calling line number in a calling number memory **30**. The central office **24** is operative to connect the call via line **31** to the called party terminal **26** in a conventional manner. When ringing the called party terminal **26**, but prior to connecting the call, the central office **24** delivers a multiple data message comprising partial calling line information **34**. The partial calling line information is delivered via line **33** to a display device **32** associated with the called party terminal **26**. It should be noted that the lines **31** and **33** in this embodiment represent different connections to the same telephone line. In alternative embodiments, the lines **31** and **33** may represent separate signal lines. For example, the line **33** could be implemented as an output signal line of the called party terminal **26**. The display device **32** in this exemplary embodiment includes a display **36** which displays the partial calling line information to the called party. The display device **32** may be a liquid crystal display (LCD) or other type of display combined with the called party terminal **26**, a personal computer terminal, a television set or other video display. The present invention may also be used with a speaker or other audio display or any other display device capable of presenting partial calling line information to a called party.

As noted previously, a telephone network providing a conventional caller ID service generally delivers the entire NPA-NXX-XXXX calling line number to the called party prior to connecting a given call, and can therefore unduly infringe the privacy rights of the caller. The conventional solution to this privacy problem allows the caller to instruct the network to block completely the delivery of calling line information to called parties, which substantially reduces the benefit of the calling line number delivery service to called parties. The present invention provides a better solution by allowing the network to deliver partial calling line information to the called party when the caller has requested privacy. The complete calling line number is not delivered, so the privacy of the caller is adequately protected. The partial information is useful to the called party and provides much of the benefit of unrestricted calling line number delivery without the privacy concerns of unrestricted delivery. It should be noted that the present invention may be utilized in conjunction with rather than as a replacement for a conventional caller ID service. That is, a system configured in accordance with the invention could deliver a full calling line number as well as other partial calling line information for callers that have not requested privacy, and could deliver only the partial calling line information for callers that have requested privacy.

The following partial calling line information **34** may be displayed on display device **32** in accordance with the present invention. It should be noted that legal restrictions may limit the specific types of calling line information which may be delivered in a given application.

(1) NPA-NXX portion of an NPA-NXX-XXXX calling line number;

(2) state, county, city, township, telephone exchange name, zip code and other geographic information associated with the calling line;

(3) operator assisted call indication;

(4) collect call indication;

(5) toll/local call indication; and

(6) indication that the call originates from:
- a residence
- a business, and if so whether a Centrex, PBX or OUTWATS
- a pay phone
- a cellular telephone
- a hotel/motel
- a hospital
- a prison

These are only examples of the types of partial calling line information which may be delivered in accordance with the invention. Numerous other types of information indicative of an incoming call characteristic may also be displayed. The state, county, city, township, telephone exchange name, zip code and other geographical information associated with the calling line number may be obtained from a table look-up using the NPA-NXX as the search key.

The above-listed partial calling line information provides valuable assistance to a called party deciding how to handle a given incoming call. For example, called parties may be more likely to answer calls from public pay phones since these are often more urgent and in general cannot be returned at a later time. Calls from mobile cellular telephones and operator assisted calls may also indicate a sense of urgency. Called parties may be more likely to answer a call from a hotel or motel, particularly if a family member, close friend or business associate is out of town. Called parties may also give preference to long-distance calls since such calls indicate a rather interested party and returning the calls may be expensive. Called parties may be able to guess who is calling by recognizing a familiar distant NPA-NXX. Residential called parties may wish to skip business calls at night, unless from a familiar NPA-NXX, and may particularly want to avoid OUTWATS calls originating from businesses. Given that called parties typically receive a large number of calls from unfamiliar calling line numbers, the partial calling line information delivered in accordance with the present invention may be more valuable in many situations than the complete calling line number.

FIG. **3** shows another exemplary embodiment of the invention. A call initiated by a caller from caller terminal **12** is routed through originating central office **14** to destination central office **24** via switching network **16**. The originating central office **14** provides the calling line number and other calling line information such as class of service information to the destination central office **24** using the above noted SS7 data transmission capability of the switching network **16**. For example, the calling line information may be supplied as part of an otherwise conventional Automatic Number Identification (ANI) sequence used for in-band signaling, or as part of the originating line information parameter octet of SS7. Alternatively, the calling line information or portions thereof could be obtained by the destination central office

through a query to an intelligent network (IN) node such as a service control point (SCP) or a network adjunct unit, or by table look-up in memory **30** of the central office **24**.

The destination central office **24** stores the full calling line number in memory **30** as previously noted. When ringing the caller terminal **26**, but prior to the connection of the call, the destination central office **24** sends the partial calling line information to display device **32** as in the previous embodiment. The central office also sends a single data message to a second display device **40** via line **41**. The single data message may be in a format similar to that used for conventional calling number delivery services. The second display device **40** in this embodiment includes a display **42** of the NPA-NXX portion of the calling line number, with the last four digits XXXX replaced with a multi-character code indicative of the type of call. In the embodiment of FIG. **3**, the multi-character code is WATS which indicates that the calling line number corresponds to a business outbound WATS or 800 service line. This informs the called party that the incoming call is from a business with a WATS line. The called party may therefore decline to accept the call, or may choose to log the call for later return knowing that the return call will be toll-free. Other exemplary multi-character codes for different call types include: RESD for residence; BUSN, PBX, CTRX or WATS for different types of business lines; COIN or PAYF for pay phones; HOTL or MOTL for hotels/motels; HOSP for hospitals; and MOBL or CELL for cellular phones. Of course, many other types of multi-character codes could also be used. It should be noted that other embodiments of the invention with more sophisticated display devices could provide more detailed displays of call type, such as full English descriptions of the call type and/or icons indicative of the call type.

The communication network of FIG. **1** may store an incoming call log for a given called party. The complete calling line numbers may be stored in the memory **30** of the destination central office **24**, or in an IN node or other element of the switching network **16**. The called party has access to the incoming call log which may include the date and time of a particular call, as well as the partial calling line information previously described. The called party can call in to the central office **24** using a predetermined central office number, and enter dual-tone multiple frequency (DTMF) commands in a conventional manner to access the incoming call log data. For example, the called party may enter the called party telephone number followed by a Personal Identification Number (PIN) code. The called party can then enter DTMF commands to retrieve specific log records and to direct the central office **24** to return a particular call from the incoming call log. The called party can therefore return an incoming call at any time and from any caller terminal by simply accessing the stored log of the central office **24** and directing a call return. The complete calling line number is stored in the central office such that the called party never has access to it. Alternative embodiments may allow a called party to access the stored call log from a personal computer or any other communication terminal. It should be emphasized that the use of DTMF commands herein is exemplary only, and that other communication channels, including without limitation the ISDN D channel, could be used to convey information between a called party terminal and the central office.

A called party may receive incoming calls from different callers having the same NPA-NXX calling line prefix, such that additional information is needed to distinguish between the callers. This may be accomplished in accordance with the invention by the called party assigning a user-selected

name to the calling party when an incoming call from that calling party is received. At any time after the incoming call is terminated, the called party can access the above-described log of incoming calls and instruct the central office **24** or other network element to associate the user-selected name with the calling party which initiated the call. The called party may enter the selected name a character at a time using conventional DTMF commands. The selected name may be the actual name of the caller or any other suitable designation such as salesman, blabbermouth, mom, dad, boss, Jimmy and the like. After the selected name has been associated with the corresponding calling line number, the central office **24** will direct the display of the selected name as partial calling line information for any subsequent calls from that calling line. Since the central office **24** does not provide the full calling line number, the privacy of the caller is protected, yet the called party can associate the calling party with previous calls. This feature is useful for distinguishing callers with the same NPA-NXX calling number prefix, as well as for screening calls from undesirable repeat callers. It should be noted that the user-selected names could also be used when full calling line numbers are delivered.

A given called party can request deletion of any record from the incoming call log or request that a record be kept until deletion is requested. Absent such requests, the stored log entries may be maintained for a predetermined time period such as 30 days after the initial call was received or 30 days after the stored record was last used to call back. As an added benefit to the called party, the network can include all incoming calls in the incoming call log. Only the partial calling line information is made available to the called party for those calls with blocked calling numbers, while the full calling line number is made available to the called party for other calls. The user can chose among several different formats for the incoming call log, using various arrangements of the date, time and partial calling line information. The stored log information may be delivered to the called party in audio or video format, depending on the type of terminal used to access the stored log information. For example, the log information may be delivered to the called party over a conventional POTS connection via a synthesized voice. The called party can signal the network via DTMF or other types of commands to skip particular entries, delete entries, dial the calling number corresponding to a particular entry, or return to a previous entry. The network could also provide printed copies of the log either periodically or upon request.

As noted above, certain of the calling line information, such as the state, county, city, township, zip code associated with the calling line, may be obtained using a table look-up in the central office **24** or other network element with the NPA-NXX prefix as a search key. The following will provide an estimate of the amount of additional memory required to provide such a feature. The NPA portion of the calling line number generally includes a digit from 2 to 9, followed by two arbitrary digits from 0 to 9, such that there are about 800 distinct NPAs. The NXX portion also includes a digit from 2 to 9 followed by two arbitrary digits, such that there can be about 800 distinct NXX codes. An exemplary geographic information look-up table may therefore include an entry for each of the about 640,000 possible NPA-NXX combinations. Each entry may include a 3-byte portion specifying zip code and state, and up to 12 ASCII characters specifying additional geographic information such as city, township, county and the like. This represents a total of about 15 bytes per entry for a total look-up table size of about 9.6 Mbytes. The look-up table can be easily accommodated using only a single conventional memory chip or a few conventional memory chips.

The following will provide an estimate of the amount of additional memory required to provide the above-described incoming call log. A large central office may serve on the order of 100,000 different telephone customers. It will be assumed that about 15% of the 100,000 customers will subscribe to the hidden calling number service described herein. Each customer may need enough space for 25 entries, including one entry for the customer identification and password, resulting in a total of 375,000 entries in the central office. Each entry may contain 5 bytes for storing a 10-digit calling number, 1 byte of line information, 2 bytes representing the date, 2 bytes representing the time, and 10 bytes for a user-selected name, for a total of 20 bytes per entry. The total memory required to provide the described call log feature in a given central office is therefore on the order of 7.5 Mbytes, which could be accommodated using only a single conventional memory chip or a few conventional memory chips.

Many of the above-described hidden calling number features can be provided without any modification to existing customer premises telephone equipment, although an improved display provided by a personal computer or other audio or video device can enhance certain of the described features. In addition, the invention can be implemented by making only minimal software alterations and memory upgrades to existing network hardware. The invention may be configured to utilize existing standard protocols and generally does not require access to remote databases.

The above-described embodiments of the invention are intended to be illustrative only. Numerous alternative embodiments may be devised by those skilled in the art without departing from the scope of the following claims.

What is claimed is:

1. A method of processing calls in a communication network, the method comprising the steps of:

receiving in a network element a call initiated by a caller over a calling line to a called party;

delivering partial calling line information to the called party prior to connecting the call, partial line information including only a first portion of a calling line number corresponding to the calling line, such that a second portion of the calling line number is hidden from the called party; and

storing the full calling line number associated with the calling line in an incoming call log maintained for the called party.

2. The method of claim 1 further including the step of storing the full calling line number with a name selected by the called party.

3. The method of claim 2 further including the step of supplying the name selected by the called party to the called party as part of the partial calling line information for a subsequent call initiated over the calling line to the called party.

4. The method of claim 1 further including the step of using the stored calling line number to initiate a return call to the calling line in response to a subsequent request from the called party.

5. An apparatus for processing calls in a communication network, the apparatus comprising:

a memory for storing calling line numbers; and

a processor coupled to the memory and operative to receive a call initiated by a caller over a given calling

line to a called party, and for delivering partial calling line information to the called party prior to connecting the call, the partial calling line information including only a first portion of a calling line number corresponding to the given calling line, such that a second portion of the calling line number is hidden from the called party;

wherein the memory stores the full calling line number associated with the calling line in an incoming call log maintained for the called party.

6. The apparatus of claim 5 wherein the memory stores the full calling line number with a name selected by the called party.

7. The apparatus of claim 6 wherein the processor is operative to supply the name selected by the called party to the called party as part of the partial calling line information for a subsequent call initiated over the calling line to the called party.

8. The apparatus of claim 5 wherein the processor is further operative to utilize the stored calling line number to initiate a return call to the calling line in response to a request from the called party.

9. A method of processing calls in a communication network, the method comprising the steps of:

receiving in a network element a call initiated by a caller over a calling line to a called party; and

delivering to the called party prior to connecting the call only a first portion of the calling line number corresponding to the calling line and delivering, in place of a second portion of the calling line number, a generalized indication of the nature of the calling line.

10. The method of claim 9 wherein said generalized indication identifies the calling line as an OUTWATTS line.

11. The method of claim 9 wherein said generalized indication identifies the calling line as a pay phone line.

12. The method of claim 9 wherein said generalized indication identifies the calling line as a cellular telephone line.

13. A method of processing calls in a communication network, the method comprising the steps of:

delivering a full calling line number as well as other partial calling line information to destination called parties of a first group of callers, wherein the first group of callers have not requested privacy; and

delivering only the partial calling line information to destination called parties of a second group of callers that have requested privacy.

14. The method of claim 13 wherein the full calling line number is of the form NPA-NXX-XXXX and the step of delivering only the partial calling line information to the destination called parties includes delivering the NPA-NXX portion such that the XXXX portion remains hidden from the destination called parties.

15. The method of claim 13 wherein the partial calling line information delivered to the destination called parties further includes an indication of a characteristic of the calling line.

* * * * *

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United States Patent [19]

Cave et al.

[11] **Patent Number:** 5,740,233

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[54] **SYSTEM AND METHOD FOR STATISTICAL DIAGNOSIS OF THE OPERATION OF AN AUTOMATED TELEPHONE SYSTEM**

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[21] Appl. No.: **556,813**

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[52] **U.S. Cl.** **379/113**; 379/134; 379/139; 379/265

[58] **Field of Search** 379/34, 112, 113, 379/133, 134, 135, 136, 137, 139, 265, 266, 309

[56] **References Cited**

U.S. PATENT DOCUMENTS

5,214,688 5/1993 Szlam et al. 379/112
5,425,087 6/1995 Gerber et al. 379/113
5,465,286 11/1995 Clare et al. 379/34

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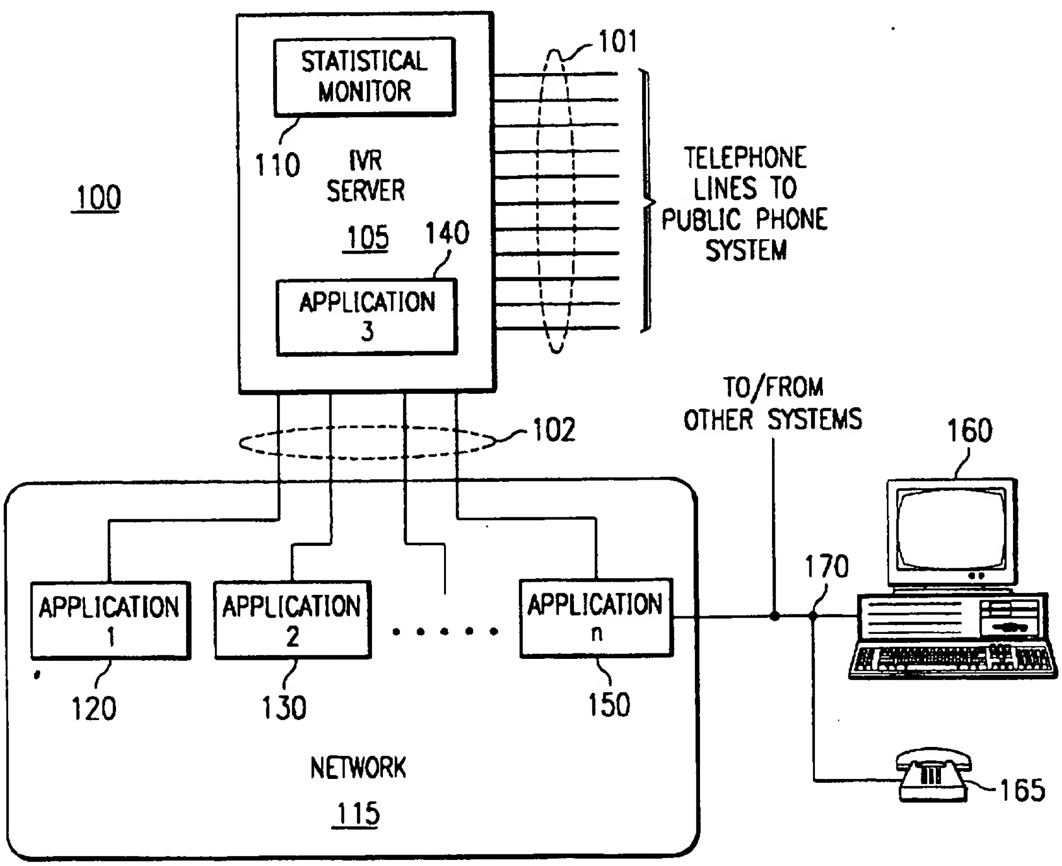
[57] **ABSTRACT**

There is disclosed a system and method for monitoring the performance of a multi-line automated telephone system and detecting complete or intermittent faults on the incoming phone lines and in interactive applications accessed by the automated telephone device. The system gathers and analyzes statistical parameters associated with each incoming phone line and compares the parameters with those of other phone lines and with past statistical averages to detect out-of-tolerance lines. The system also gathers and analyzes statistical parameters associated with the execution of various applications and compares the parameters with past statistical averages to detect out-of-tolerance accesses to the applications.

50 Claims, 3 Drawing Sheets



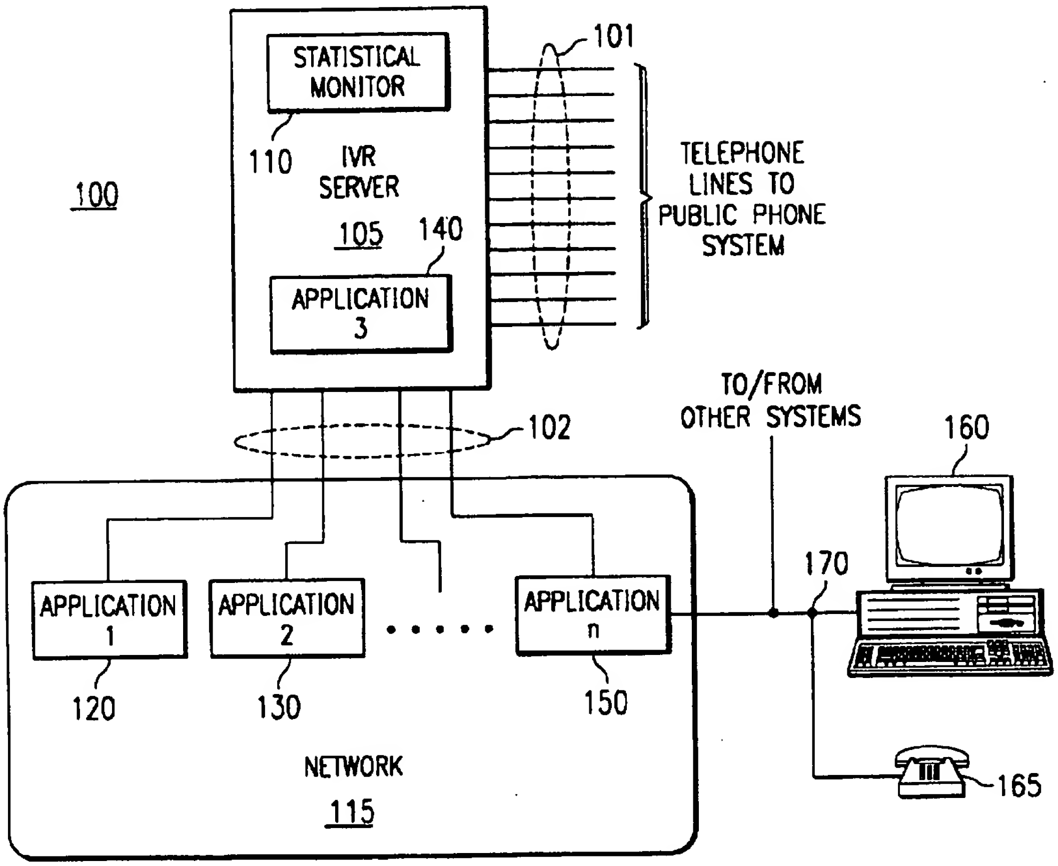


FIG. 1

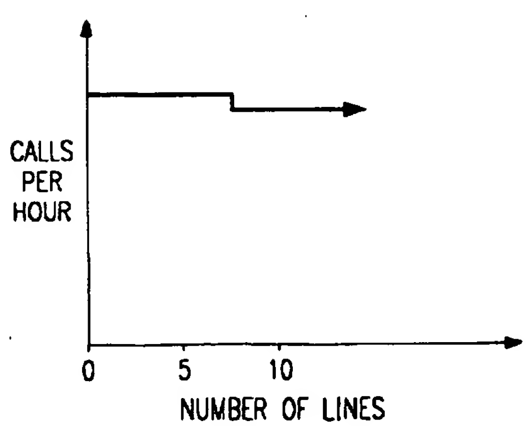


FIG. 3

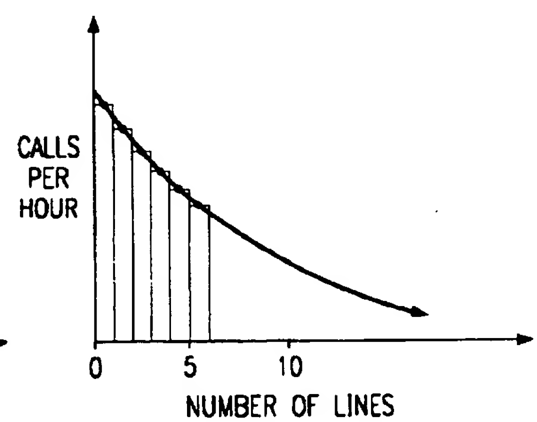


FIG. 4

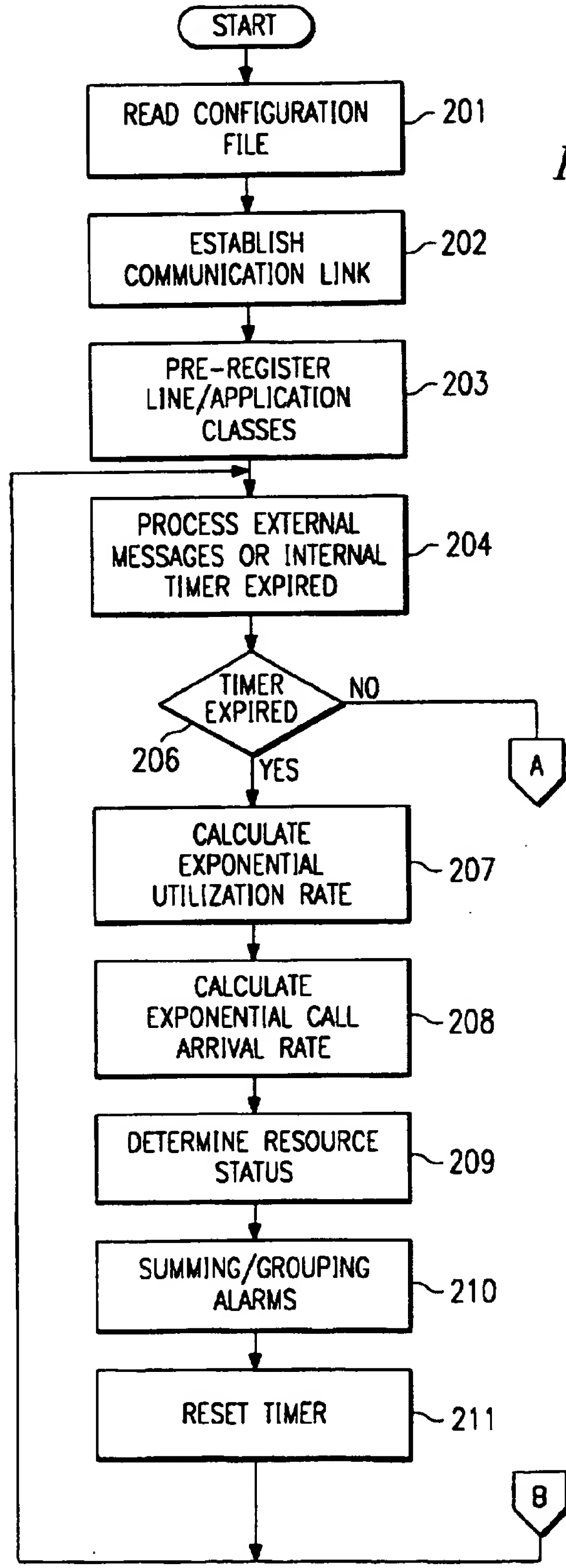

START
READ CONFIGURATION FILE
201
ESTABLISH COMMUNICATION LINK
202
PRE-REGISTER LINE/APPLICATION CLASSES
203
PROCESS EXTERNAL MESSAGES OR INTERNAL TIMER EXPIRED
204
TIMER EXPIRED
NO
206
YES
A
CALCULATE EXPONENTIAL UTILIZATION RATE
207
CALCULATE EXPONENTIAL CALL ARRIVAL RATE
208
DETERMINE RESOURCE STATUS
209
SUMMING/GROUPING ALARMS
210
RESET TIMER
211
B

FIG. 2A

200

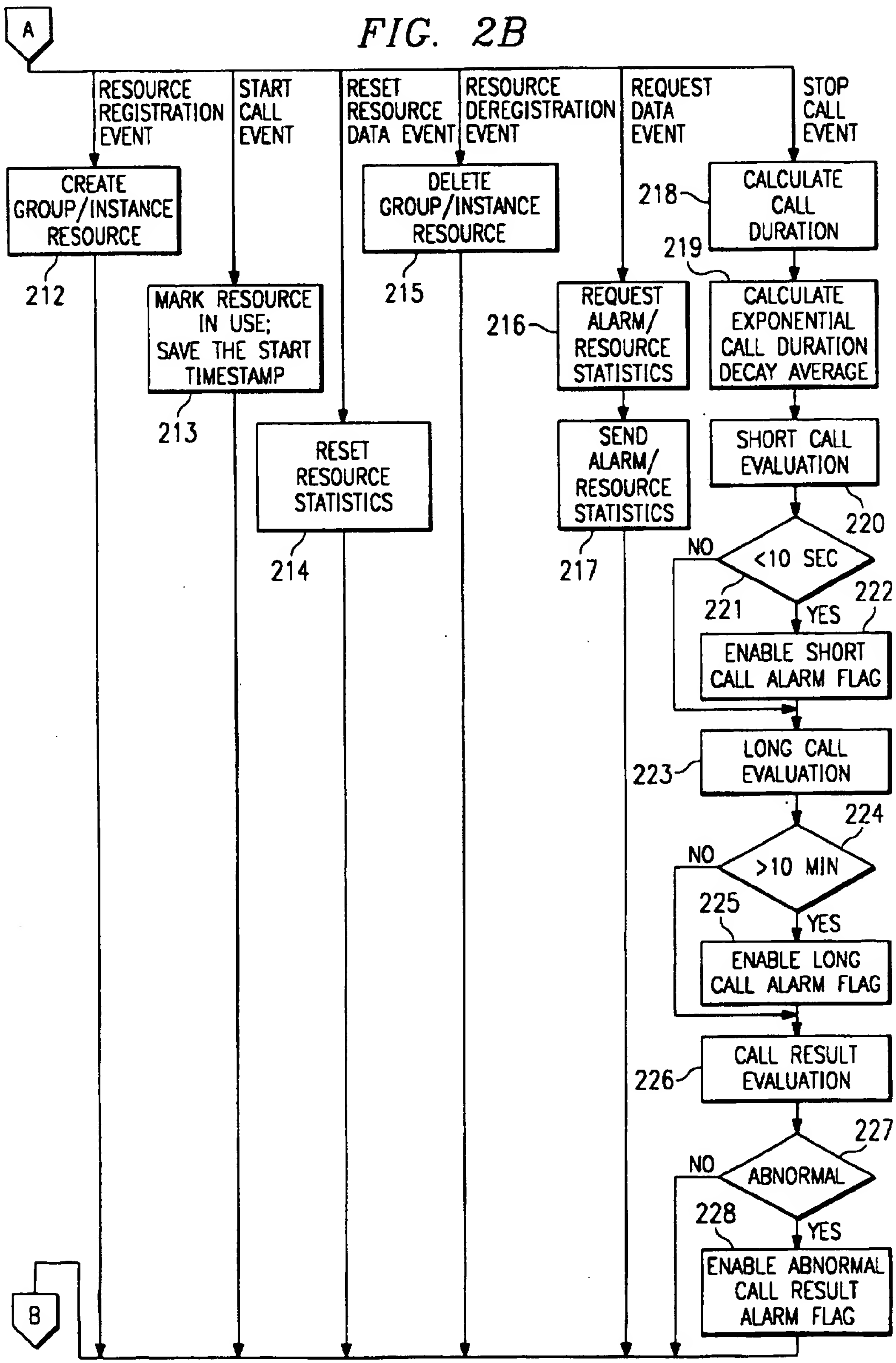


FIG. 2B

01/28/2004, EAST Version: 1.4.1

SYSTEM AND METHOD FOR STATISTICAL DIAGNOSIS OF THE OPERATION OF AN AUTOMATED TELEPHONE SYSTEM

TECHNICAL FIELD OF THE INVENTION

This invention relates to a device for monitoring and diagnosing the performance of a multi-line automated telephone device, and in particular to a device for monitoring and diagnosing the operation of multiple telephone lines and interactive applications in an interactive voice response (IVR) environment.

BACKGROUND OF THE INVENTION

Automated telephone systems employing auto attendant features and/or interactive voice response applications are becoming increasingly popular as more and more businesses realize these systems' potential for reducing human operator costs and for increasing the number and diversity of services available to the public. This has correspondingly caused a great increase in the capacity and complexity of automated telephone systems.

While the exact configurations of these systems vary, the most common systems typically employ an interactive voice response (IVR) unit coupled to a large switch, such as a PBX, that interfaces with a large number of incoming telephone lines from the public telephone system. The IVR system also interfaces with enterprise data through database queries, and screen emulation techniques. The interactive voice response (IVR) units are capable of playing scripted messages to incoming callers, prompting the callers to enter DTMF keypad inputs to select among various services, and recognizing and decoding the corresponding DTMF inputs received from the caller. Information from the databases is spoken to the caller. More advanced automated telephone systems can place outbound calls using predictive dialers and employ voice recognition technology to analyze a person's spoken response to a scripted prompt to determine what service that person wishes to select. Still more advanced systems are capable of communicating with other computer systems via a modem connection. Finally, still other automated telephone systems are capable of sending and receiving facsimile (fax) transmissions.

Once received, the incoming call is typically directed by the IVR unit to one or more applications selected by the caller. These applications are too numerous and diverse to fully describe here. However, the more popular applications used by most customers pertain to bank account information, credit card account information, voice mail/voice messaging, catalog ordering, stock information, investment plan information (such as 401(k) plans), or entertainment services, such as pay-per-view. Additionally, as previously noted, some applications may be facsimile transmissions or computer communications via modem.

An IVR unit is typically coupled to a computer database system via a network, usually a Local Area Network (LAN). LAN systems are popular because they allow the IVR unit to access data and applications located in remote nodes. This is particularly true in those instances where a LAN interconnects separately owned businesses that subscribe to the network in order to obtain interactive voice response services that they might not be able to afford by themselves.

There are, however, numerous problems associated with the performance and reliability of these systems. This is especially true as the number of incoming telephone lines to an IVR unit increases. Frequently, a malfunctioning line connection to an IVR unit having many incoming telephone lines will go undetected for an extended period of time. One or more lines in a multi-line system may malfunction consistently or intermittently without being noticed by the owner/operator of the automated telephone system because the remaining properly functioning lines will continue to operate and allow the automated telephone system to function.

Before going on, it might be helpful to provide definitions for a few types of possible line distribution:

First available distribution passes the next call to the lowest numbered available line.

Round-robin distribution passes the next call to the idle line number next highest than the line to which the previous call was connected. This will sequence through all of the lines and eventually restart at the lowest line number.

Longest idle distribution passes the next call to the line in the group which has been idle the longest.

There are numerous ways in which telephone line connections can be faulty. The public telephone system may simply fail to deliver incoming telephone calls on a particular telephone line. This is a fault that is beyond the control of the automated telephone system owner/operator to detect. A malfunction in the automated telephone system may cause the IVR unit to answer an incoming line and immediately hang up without interacting with the customer. The customer may then simply call back and connect to a different incoming telephone line that is properly functioning. In such a case, the malfunction causing the first line to hang up rapidly will go unnoticed unless the customer makes a complaint, which is unlikely.

Another type of error that may occur is that the telephone connection will fail to disconnect when the customer hangs up. In such a case, the line will be held open by the interactive voice response unit for an extended period of time, perhaps even hours. Other possibilities are that an incoming call on a particular line will continually receive either a busy signal or endless ringing without an answer by the interactive voice response unit, or will be noisy.

In all of the foregoing situations, the commercial business owning an IVR unit or the service provider selling IVR services to third parties will not become aware of these problems unless and until a noticeable drop-off in the capacity or through-put of the automated telephone system occurs, or customers complain that they attempted to use the system and the system rang without answering, or prematurely disconnected during the phone call. The latter case is unlikely because the customers usually call back a second time, connect with a properly functioning line, and disregard the malfunctioning connection.

Furthermore, while a consistent failure of a single line in a multi-line system may often go unnoticed for an extended period of time, it is at least possible, in most cases, to diagnose such consistent failures by running self-test programs on the automated telephone system. But, in the case of an intermittent failure of a telephone line, it is quite often impossible to detect the fault.

An additional source of error in automated telephone systems can occur in the application modules driving the interactive voice response unit. There are numerous hardware and software sources of error that can cause a particular application to fail intermittently during routine operation. These intermittent failures of applications are particularly difficult to detect because the fault does not always occur on the same incoming telephone line. Therefore, a fault with the application cannot be detected by merely trouble-shooting the telephone lines connected to the IVR unit.

This situation is particularly exacerbated in those cases where malfunctioning application modules may not be on the same system as the IVR server or an IVR service provider connects through a network to third party subscribers. The service provider typically has no control over the quality of the software and hardware used by the network subscriber. In such a case, the ability of the service provider to detect and correct a fault in the application, or alert the subscriber to the fault, is very limited.

There is therefore a need for systems and methods for verifying the operation of a multi-line automated telephone system and detecting complete and/or intermittent faults within such a system even though there are no outward manifestations of such faults.

There is a further need for systems and methods for monitoring telephone lines coupled to a multi-line automated telephone system and detecting both complete and intermittent faults in the telephone line connections.

There is a still further need in the art to provide systems and methods for monitoring the execution of application modules that interact through an interactive voice response unit and for detecting both complete and intermittent faults in the application modules.

SUMMARY OF THE INVENTION

These and other problems inherent in the prior art have been solved by providing a statistical health monitor associated with an interactive voice response (IVR) unit that compiles and analyzes statistical parameters related to the operation of telephone lines connected to the interactive voice response system. The statistical health monitor of the present invention also compiles and analyzes operational parameters associated with the execution of application modules associated with the interactive voice response unit.

In one embodiment of the present invention, a statistical engine monitors incoming telephone lines and calculates the incoming call rate of telephone calls being handled by the IVR unit in a first available distribution across multiple telephone lines connected to the public telephone system.

In another embodiment of the present invention, a statistical engine monitors incoming telephone lines and calculates the incoming call rate of telephone calls being handled by the IVR unit in a round-robin distribution service across multiple telephone lines connected to the public telephone system.

In a further embodiment of the present invention, a statistical engine monitors incoming telephone lines and calculates the number of telephone calls falling below a minimum threshold duration or exceeding a maximum threshold duration on a line-by-line basis across multiple telephone lines.

In all of the foregoing embodiments, the key parameters monitored by the present invention are call arrival rate and the duration of each call. The present invention detects intermittent faults on a particular line by comparing the statistical parameters of that line with the preceding and/or following telephone lines, or against any other or all other telephone lines in both rotary distribution and in sequential distribution service. Furthermore, the present invention can diagnose intermittent faults in a particular telephone line by comparing the statistical parameters of that line with the statistical parameters of a corresponding line in a "model distribution" for either a rotary distribution or a sequential distribution service.

It is a technical advantage of the present invention that a statistical engine monitors the instances of each application module accessed by the IVR unit, both locally and through a network, and calculates the operational parameters associated with those application modules. The statistical health monitor of the present invention records the start time and stop time of each execution of an application module and develops a usage profile for each application module based on the average duration of the accesses to each application module. If the average duration of each access to an application module increases or decreases dramatically, the present invention diagnoses the change as a fault and sets an alarm or flag accordingly.

It is another technical advantage of the present invention that it also detects application module accesses that fall below a minimum threshold duration or exceed a maximum threshold duration and sets an alarm or flag accordingly.

The foregoing has outlined rather broadly the features and technical advantages of the present invention in order that the detailed description of the invention that follows may be better understood. Additional features and advantages of the invention will be described hereinafter which form the subject of the claims of the invention. It should be appreciated by those skilled in the art that the conception and the specific embodiment disclosed may be readily utilized as a basis for modifying or designing other structures for carrying out the same purposes of the present invention. It should also be realized by those skilled in the art that such equivalent constructions do not depart from the spirit and scope of the invention as set forth in the appended claims.

BRIEF DESCRIPTION OF THE DRAWINGS

For a more complete understanding of the present invention, and the advantages thereof, reference is now made to the following descriptions taken in conjunction with the accompanying drawings, in which:

FIG. 1 is a system block diagram of an interactive voice response telephone system employing a statistical health monitor in accordance with the present invention;

FIGS. 2A and 2B are a logical flow diagram of the operations of the present invention;

FIG. 3 is a graph of the call distribution of incoming calls across multiple phone lines in a round-robin distribution phone system; and

FIG. 4 is a graph of the call distribution of incoming calls across multiple phone lines in a first available distribution phone system.

DETAILED DESCRIPTION OF THE INVENTION

FIG. 1 depicts an automated telephone system **100** coupled to the public telephone system through a group of telephone lines **101**. Incoming telephone lines **101** are connected to interactive voice response server **105**. Interactive voice response (IVR) server **105** is coupled by connections **102** to computer network **115**. Associated with interactive voice response (IVR) server **105** is statistical health monitor **110** in accordance with the present invention. It may or may not be in the box. Incoming calls received on telephone lines **101** may access Application 1 through Application N, shown as items **120**, **130**, **140** and **150**, through IVR server **105**. These applications may include access to a human telephone agent coupled by connection **170** to network **115**. Typically, the telephone agent uses computer terminal **160** and voice telephone **165** to communicate with a caller on telephone lines **101**.

As shown in FIG. 1, Applications 1–N can be distributed at numerous nodes throughout network **115** or may be

resident within IVR server 105. Additionally, a telephone agent using voice telephone 165 and computer terminal 160 may be coupled to IVR server 105 through network 115, as shown in FIG. 1, or may be coupled directly to IVR server 105 (not shown).

Statistical health monitor 110 has the ability for monitoring telephone lines 101 to detect the initiation of an incoming telephone call on any of telephone lines 101 and to detect the termination of each said incoming call. Statistical health monitor 110 also has the ability for recording the start time and stop time of each incoming call received on telephone lines 101, calculating the duration of each said call, and calculating and maintaining other statistical parameters on a line-by-line basis for each one of telephone lines 101. Statistical parameters are maintained for each line, including the total number of calls received on each line, individual call duration, average call duration, and call arrival rate (i.e., number of calls per unit of time).

Statistical health monitor 110 also contains circuitry for monitoring each access to an application by IVR server 105 and for calculating statistical parameters associated with the applications on an application-by-application basis. Statistical health monitor 110 records the start time and stop time of each execution of an application, calculates the duration of each access to the application, calculates and updates the average duration of the accesses to a particular application.

Before discussing the logical flow diagram illustrated in FIGS. 2A and 2B, attention is drawn to FIGS. 3 and 4 which illustrate the distribution of incoming telephone calls in a rotary (round robin) distribution and a first available distribution, respectively. Incoming telephone calls may be received from the public telephone company in either of these modes, as selected by the customer.

In the rotary distribution illustrated in FIG. 3, incoming calls are directed to individual telephone lines in an evenly rotating manner, without giving priority to particular lines. The first incoming telephone call is directed to Line 1, the second incoming call is directed to Line 2, the third incoming call is directed to Line 3, and so forth, until the last telephone line in the multi-line system has received a call. The next incoming call is directed to by Line 1, if available. If Line 1 is not available, the incoming call will be sent to the next available line in ascending order.

In a rotary distribution to an automated telephone system having thirty incoming telephone lines, the first thirty incoming calls received will be distributed from Line 1 to Line 30 in ascending order. Assuming that each incoming telephone call is relatively short in duration and that the incoming calls are spaced relatively far apart. Line 1 will be idle when the 31st incoming call is received and the following incoming calls will continue to be distributed in order to Line 2, Line 3, Line 4, etc. In this manner, incoming calls are distributed evenly across all incoming telephone lines.

If the incoming calls have an arrival rate of 300 calls per hour, then each line receives 10 calls per hour (still assuming 30 lines). If incoming calls are received at a high rate and a particular line is occupied by a single call for an extended period of time, that line will be skipped if it is still busy when its next turn comes around in the rotary distribution. However, over an extended period of time, long duration calls will be randomly, and therefore evenly, distributed across all of the incoming telephone lines and the call distribution will be approximately even across all telephone lines, as shown in FIG. 3.

The alternative service to rotary distribution is sequential distribution, wherein incoming calls are directed to telephone lines of the highest priority first, if those lines are available. In a sequential distribution, the first incoming call is directed to Line 1, and the following incoming calls are directed to the other telephone lines in ascending numerical order, provided that no telephone line having a lower numerical value is idle.

For example, if Line 1 receives the first incoming telephone call and the next two calls are received while Line 1 is still busy, the next two incoming calls are directed to Line 2 and Line 3, respectively. However, if a fourth incoming call is received after Line 1 has become idle, the fourth incoming call will be directed to Line 1, not Line 4. If a fifth incoming call is received while Lines 1–3 are busy, the fifth incoming call will be directed to Line 4. If, however, Line 1, Line 2, or Line 3 has become available when the fifth incoming call is received, the fifth incoming call will be directed to the now available line, and not to Line 4. In this manner, the distribution of incoming telephone calls is heaviest on Line 1 and lightest on Line 30 in a thirty line automated telephone system.

An example of the distribution of incoming telephone calls in a sequential distribution is shown in FIG. 4, which has the characteristic of a decaying exponential curve. The rate at which the decaying exponential curve drops off depends upon the average duration of each telephone call and the arrival rate of incoming telephone calls.

If the average duration of each telephone call is very short and the arrival rate of incoming calls is low, the first few lines will be able to handle all of the incoming calls and the exponential curve will have a sharply decreasing slope. Line 1 will handle the greatest number of calls, with a sharp drop off to Line 2, which will in turn have a sharp drop off to Line 3. Lines 4 and above will rarely, if ever, be needed when the call arrival rate is very low and the average duration of each call is short.

If, however, the call arrival rate is very high and the average duration of each call is comparatively long, then more calls will be distributed to higher numerical lines and the downward slope of the decaying exponential curve will be much more gradual.

In either a round-robin distribution or a first available distribution, if all thirty lines are busy when a thirty-first call is received, the thirty-first incoming telephone call will be placed on hold by automated telephone system 100.

It should be noted that the decaying exponential curve in FIG. 4 may be readily calculated using well known math algorithms whenever both the call arrival rate and the average call duration rate are known. Thus, it is possible to predict the number of calls per hour received on any line in the multi-line system whenever the call arrival rate and the average call duration are known. As will be seen below, this ability to estimate the call volume on each line may be used to diagnose the health of each available line.

Returning now to FIGS. 2A and 2B, the logical flow diagram of the statistical health monitor of the present invention is illustrated. In FIGS. 2A and 2B, applications and telephone lines are referred to collectively as "resources." The algorithms involved in developing statistical parameters to diagnose the operations of the incoming telephone lines and the applications accessed by IVR server 105 are largely the same. Both can therefore be addressed in a single flow chart, as shown in FIGS. 2A and 2B. Resources are designated by class as either a line or an application. Within each class, there may be groups. For example, a group may consist of one or more trunk lines and the group may be either round-robin distribution lines, first available

sequential distribution lines, or longest idle distribution lines where IVR server 105 receives both types of distributions.

Statistical health monitor 110 determines the health of IVR server 105 by determining (1) the health of each line; and (2) the health of each application. To determine the health of each incoming telephone line, statistical health monitor 110 determines the normal operation of the incoming telephone lines from the public telephone system and the corresponding normal distribution of incoming calls. Statistical health monitor 110 then develops statistical parameters for line utilization, call duration and call arrival rate. Statistical monitor 110 evaluates individual call durations in order to set short call alarms and long call alarms, and counts incoming calls per unit time in order to set abnormal call arrival rate alarms. Statistical monitor 110 may also receives call result error codes from the public telephone system which indicate faults occurring on particular lines during outbound and inbound calls.

In determining the health of an application, statistical health monitor 110 evaluates individual application access durations in order to set short access duration alarms, long access duration alarms, and an abnormal access duration average alarm. Additionally, statistical monitor 110 receives application result error codes from the applications and from network 115 whenever an error occurs in the execution of an application.

Upon initialization, statistical health monitor 110 obtains configuration information associated with IVR server 105 in order to determine the number of incoming telephone lines, the type of call distribution, and the number and type of applications accessible to IVR server 105. Statistical health monitor 110 then determines the incoming line and application classes (Steps 201–203 in FIG. 2A). Next, statistical health monitor 110 processes any external error messages that have been received and verifies whether or not the internal timer has expired on a resource event (Steps 204 and 205 in FIG. 2A).

Four events are required to be received in order to calculate the statistical parameters: (1) Resource Registration Event, (2) Resource Deregistration Event, (3) Resource Start Event, and (4) Resource Stop Event. (Steps 212, 213, 215, and 218 in FIG. 2B.)

A Resource Registration Event informs statistical health monitor 110 that a new resource (telephone line or application) needs to be added to the calculations. Resource Registration Events include an Event ID, Resource Type, and Resource ID.

A Resource Deregistration Event informs statistical health monitor 110 that a resource needs to be removed from the calculations. Resource Deregistration Events include an Event ID, Resource Type, and Resource ID.

A Resource Start Event informs statistical health monitor 110 that a resource is allocated. Resource Start Events include an Event ID, Resource Type, Resource ID, and Start TimeStamp.

A Resource Stop Event tells statistical health monitor 110 when a resource is released. Resource Stop Events include an Event ID, Resource Type, Resource ID, Start TimeStamp, Stop TimeStamp, and a Result Code.

There are three principal statistical parameters used to determine the health conditions of a system resource. These statistical parameters are: utilization average, call/access duration, and call arrival (usage) rate. All statistical parameters calculated by the present invention use an exponentially decaying average method. The exponential decay average behaves like a windowed average, with a δ value determining the equivalent window size. The benefit of an exponential decay average is that large amounts of historical data can be automatically dropped off and recent data are the most heavily weighted. The following is a brief explanation of the algorithms involved in calculating some of the relevant statistical parameters.

The duration average, A(t), is the average amount of time that the resource (line or application) has been in use. A(t) is given by:

$$A(t)=\delta*A(t-1)+(1-\delta)*X$$

where A(t−1) is the last average value, A(t) is the new average value, X is the resource in-use time in seconds, T is the sampling time interval value, and δ is the window size factor (default=0.95). For the case A(0), the initial value will be set to the first exponential call/access duration result and the next three call/access duration results will use δ=0.50 to perform the calculation.

The utilization average, A(t)/T, is the percentage of the time that the resource has been in use. A(t) is given by:

$$A(t)=\delta*A(t-1)+(1-\delta)*L(t)$$

where A(t−1) is the last average value, A(t) is the new average value, L(t) is the time in use after the last interval, T is the sampling time interval value, and δ is the window size factor (default=0.95). For the case A(0), the initial value will be set to the first exponential average result and the next three results will use δ=0.50 to perform the calculation.

The call arrival (usage) rate average, A(t)*3600/T, is a measure of how frequently a line is put into use per hour and is given by:

$$A(t)=\delta*A(t-1)+(1-\delta)*W$$

where A(t−1) is the last average value, A(t) is the new average value, W is the number of calls received after the last interval, T is the sampling time interval value, and δ is the window size factor (default=0.95). For the case A(0), the initial value will be set to the first exponential call arrival rate result, and the next three results will use δ=0.50 to perform the calculation.

Steps 218–228 of FIG. 2B illustrate the sequence of calculations performed by statistical health monitor 110. In Steps 218–228, a "call" may be an actual telephone call received on a particular incoming telephone line, or an "access" to an application by IVR server 105. For the purpose of simplicity, "call" is used for both application accesses and incoming telephone calls.

Statistical health monitor 110 maintains a "result window" covering the last ten results received by statistical health monitor 110 for a particular incoming telephone line, or a particular application. If the current result of call duration on a specific telephone line is too short (less than 10 seconds) then statistical health monitor 110 gives a short call alarm for that line. If the current result of call duration on a specific call is too long (more than 10 minutes), and three such occurrences have occurred within the last ten results, statistical health monitor 110 generates a long call alarm for that specific telephone line. Furthermore, if the call duration result is too long (more than 10 minutes) and is the second of two consecutive call durations in excess of 10 minutes, statistical health monitor 110 generates a long call alarm for that telephone line.

In a similar manner, statistical health monitor 110 can generate a short call alarm for an application whenever three call results are too short (less than 10 seconds) during the

last 10 results in the moving result window. The statistical health monitor **110** can generate a long call alarm whenever three call results are too long (more than 10 minutes) during the last ten results in the moving result window across a group of monitored lines. Statistic health monitor **110** also generates a long call alarm for an application whenever two consecutive accesses to an application exceed more than 10 minutes across a group of monitored lines.

Statistical health monitor **110** also receives result codes from IVR server **105** and telephone lines **101**. Statistical health monitor **110** generates call result error alarms for both telephone lines and applications whenever the current result of a system call result returns an error for the result code and three such errors have occurred within the last ten results.

Statistical health monitor **110** also generates an alarm if the call arrival rate on a specific line is significantly different from its adjacent active lines in either a rotary distribution or a sequential distribution. Depending on the exponential weight used and the amount of activity on telephone lines **101**, the tolerance for a particular telephone line may vary from +/−50% of the two adjacent active line averages or as little as a few percent of the two adjacent active line averages.

It is possible to compare the call arrival rate on a particular telephone line with the two adjacent active lines in both a rotary and a sequential distribution. In a rotary distribution, all telephone lines are expected to have approximately the same number of calls per hour. Therefore, taking the average of the preceding and following active lines (which should be approximately equal) will yield the average for all of the incoming telephone lines, including the particular line in consideration.

In a sequential distribution, the preceding line and the following line are expected to have call arrival rates higher and lower, respectively, than the line in the middle. However, since the call arrival rate on the middle line is generally close to the midpoint between the higher value on the preceding line and the lower value on the following line, averaging the preceding and following lines will yield an average close to the value expected to be found on the middle line. Therefore, in round-robin distributions, the expected value on a particular line may be roughly calculated by taking the average of the call arrival rates on all lines. The expected value for a particular line in first available distributions is calculated by taking the average of the call arrival rates on the preceding and following lines.

As noted previously, in the case of a first available distribution, it is possible to estimate the expected call arrival rate on any line in multi-line system whenever the call arrival rate and the average call duration are known. If the call arrival rate and average call duration are known, it is possible to derive a "model" first available distribution as shown in FIG. 4. The actual measured call arrival rates on each incoming telephone line in the telephone system are then compared to the estimated value in the "model" distribution.

Once statistical health monitor **110** determines that an incoming telephone line or an application is out-of-tolerance and generates a corresponding alarm, statistical health monitor **110** can relay the alarm to the owner/operator or maintenance provider of IVR server **105**. The alarm may be indicated by a simple indicator light on IVR server **105** or may be a data message transmitted to a computer terminal coupled to IVR server **105** directly or through network **115**. Alternatively, statistical health monitor **110** may cause IVR server **105** to generate an outbound telephone call on telephone lines **101** to a remotely located computer terminal or a voice message service.

In addition to diagnosing faults by comparing call arrival rates of adjacent lines and counting long calls and short calls occurring within a fixed-size call window, statistical health monitor **110** can, in certain cases, compare statistical parameter averages of telephone lines **101** (whether adjacent or not) to diagnose faults. If automated telephone system **100** serves only a single type of application (such as bank account balance information), it is reasonable to expect that all calls received by automated telephone system **100** will be tolerably close to a specified average duration. The same may also be true if automated telephone system **100** serves a limited number of different applications that nonetheless have similar average call durations. In either of these relatively limited cases, statistical health monitor can diagnose faults on telephone lines **101** by comparing individual line averages of line utilization and call duration with the composite averages for all telephones lines **101** in the automated telephone system **100**.

The same type of problem exists on outbound calls when they are part of an IVR. For example, calls can be placed and experience early termination, or non-completion, or a number of other malfunctions. This can be determined by a statistical measure against a norm, as discussed above. Also the detection of errors can be based on a statistical model which may be modified based on measured parameters or anticipated different parameters, all under the control of the system.

Although the present invention and its advantages have been described in detail, it should be understood that various changes, substitutions and alterations can be made herein without departing from the spirit and scope of the invention as defined by the appended claims.

What is claimed is:

1. An apparatus for diagnosing the operation of an automated telephone system coupled to a plurality of telephone lines, said apparatus comprising:

line monitoring apparatus for detecting an initiation and a termination of each call on each of said telephone lines coupled to said automated telephone system, measuring a duration of each said call, and counting a total number of calls on a line-by-line basis; and

a processor coupled to said line monitoring circuitry for 1) calculating a first selected statistical parameter for a first selected one of said telephone lines, and 2) comparing said first selected statistical parameter to a selected average of a corresponding second selected statistical parameter of a second selected telephone line and a corresponding third selected statistical parameter of a third selected telephone line, and 3) generating a system alarm corresponding to said first selected telephone line whenever a difference between said first selected statistical parameter and said selected average of said corresponding second and third selected statistical parameters exceeds a maximum tolerance value.

2. The apparatus as set forth in claim **1** wherein said second selected telephone line and said third selected telephone line precede and follow, respectively, said first selected telephone line in a predetermined distribution order of said incoming calls.

3. The apparatus as set forth in claim **2** wherein said predetermined distribution order is a first available distribution order.

4. The apparatus as set forth in claim **2** wherein said predetermined distribution order is a round-robin rotary distribution order.

5. The apparatus as set forth in claim **1** wherein at least one of said averages calculated by said control processor is a weighted average.

6. The apparatus as set forth in claim 1 wherein said predetermined distribution order is longest idle.

7. The apparatus as set forth in claim 1 wherein said first selected statistical parameter is a call arrival rate average of said first selected telephone line.

8. The apparatus as set forth in claim 7 wherein said call arrival rate average is given by the formula

$$A(t)=\delta*A(t-1)+(1-\delta)*W$$

where A(t−1) is the last average value, A(t) is the new average value, W is the number of calls received after the last interval, T is the sampling time interval value, and δ is the window size factor (default=0.95).

9. The apparatus as set forth in claim 1 wherein said first selected statistical parameter is a call duration average of said first selected telephone line.

10. The apparatus as set forth in claim 9 wherein said call duration average is given by the formula

$$A(t)=\delta*A(t-1)+(1-\delta)*X$$

where A(t−1) is the last average value, A(t) is the new average value, X is the resource in-use time in seconds after the last interval, T is the sampling time interval value, and δ is the window size factor (default=0.95).

11. The apparatus as set forth in claim 1 wherein said first selected statistical parameter is a line utilization average of said first selected telephone line.

12. The apparatus as set forth in claim 11 wherein said line utilization average is given by the formula

$$A(t)=\delta*A(t-1)+(1-\delta)*L(t)$$

where A(t−1) is the last average value, A(t) is the new average value, L(t) is the time in use after the last interval, T is the sampling time interval value, and δ is the window size factor (default=0.95).

13. The apparatus as set forth in claim 1 wherein said processor sends said system alarm to a remote monitor.

14. The apparatus as set forth in claim 13 wherein said processor sends said calculated data to a remote monitor.

15. The apparatus as set forth in claim 1 wherein said control processor sends said system alarm to a remote station by causing said automated telephone system to initiate an outbound telephone call to said remote station.

16. The apparatus as set forth in claim 1 wherein said processor sends said calculated data to a remote monitor for comparison of like calculated data from at least one other system prior to generating said system alarm.

17. The apparatus as set forth in claim 1 wherein said calls are incoming to said telephone system.

18. The apparatus as set forth in claim 1 wherein said calls are outgoing from said telephone system.

19. An apparatus for diagnosing the operation of an automated telephone system coupled to a plurality of telephone lines, said apparatus comprising:

line monitoring circuitry for detecting an initiation and a termination of each incoming call to said telephone system on each of said telephone lines, measuring a duration of each said incoming call, and counting a total number of incoming calls received on a line-by-line basis;

means for comparing each said duration of each said incoming call on a first selected telephone line to a maximum threshold duration and to a minimum threshold duration; and

means for generating a signal if more than a predetermined number of the last N incoming calls fall below said minimum threshold duration or exceed said maximum threshold duration.

20. The apparatus as set forth in claim 19 wherein said signal generating means further comprises means for generating a short call alarm if said predetermined number is three which fall below said minimum threshold duration.

21. The apparatus as set forth in claim 19 wherein said signal generating means further comprises means for generating a long call alarm if more than three of the last ten incoming calls exceed said maximum threshold duration.

22. The apparatus as set forth in claim 19 wherein said signal generating means further comprises means for generating a call result error alarm if two consecutive incoming calls exceed said maximum threshold duration.

23. The apparatus as set forth in claim 19 further including means for sending said signal to a computer terminal coupled to said automated telephone system.

24. The apparatus as set forth in claim 19 further including means for sending said signal to a remote station by causing said automated telephone system to initiate an outbound telephone call to said remote station.

25. An apparatus for diagnosing the operation of an automated telephone system coupled to a plurality of telephone lines, wherein incoming call are received in a sequential distribution across said plurality of telephone lines, said apparatus comprising:

line monitoring circuitry for detecting an initiation and a termination of each incoming call on each of said telephone lines, measuring a duration of each said incoming call, and counting a total number of incoming calls received on a line-by-line basis; and

a control processor coupled to said line monitoring circuitry for 1) calculating a system call duration average for all said incoming calls, 2) calculating a system call arrival rate for all said incoming calls, 3) calculating an expected call distribution of said incoming calls on each of said telephone lines based on said system call duration average and said system call arrival rate, 4) comparing a first expected call distribution of a first selected telephone line to an actual call distribution of incoming calls on said first selected telephone line and 4) generating a system alarm corresponding to said first selected telephone line if a difference between said first expected call distribution and said actual call distribution exceeds a maximum value.

26. The apparatus as set forth in claim 25 wherein at least one of said averages calculated by said control processor is a weighted average.

27. The apparatus as set forth in claim 25 wherein said control processor sends said system alarm to a computer terminal coupled to said automated telephone system.

28. The apparatus as set forth in claim 25 wherein said control processor sends said system alarm to a remote station by causing said automated telephone system to initiate an outbound telephone call to said remote station.

29. A method of diagnosing the operation of an automated telephone system coupled to a plurality of telephone lines, the method comprising the steps of:

monitoring the plurality of telephone lines to detect an initiation and a termination of each incoming call on each of the telephone lines;

measuring the duration of each of the incoming calls;

counting a total number of incoming calls received on each line;

calculating a first selected statistical parameter for a first selected one of the telephone lines;

comparing the first selected statistical parameter to a selected average of a corresponding second selected statistical parameter of a second selected telephone line and a corresponding third selected statistical parameter of a third selected telephone line, wherein the second selected telephone line and the third selected telephone line precede and follow, respectively, the first selected telephone line in a predetermined distribution order of the incoming calls; and

generating a system alarm whenever a difference between the first selected statistical parameter and the selected average of the corresponding second and third selected statistical parameters exceeds a maximum tolerance value.

30. The method as set forth in claim 29 wherein the predetermined distribution order is a sequential distribution order.

31. The method as set forth in claim 29 wherein the predetermined distribution order is a rotary distribution order.

32. The method as set forth in claim 29 wherein the first selected statistical parameter is a weighted average.

33. The method as set forth in claim 29 wherein the first selected statistical parameter is a call arrival rate of the first selected telephone line.

34. The method as set forth in claim 29 wherein the first selected statistical parameter is a call duration average of the first selected telephone line.

35. The method as set forth in claim 29 wherein the first selected statistical parameter is a line utilization average of the first selected telephone line.

36. The method as set forth in claim 29 further including the step of sending the system alarm to a computer terminal coupled to the automated telephone system.

37. The method as set forth in claim 29 further including the step of sending the system alarm to a remote station by causing the automated telephone system to initiate an outbound telephone call to the remote station.

38. A method of diagnosing the operation of an automated telephone system coupled to a plurality of telephone lines, the method comprising the steps of:

monitoring the plurality of telephone lines to detect an initiation and a termination of each incoming call on each of the telephone lines;

measuring the duration of each of the incoming calls;

counting a total number of incoming calls received on each line;

comparing each duration of each incoming call on a first selected telephone line to a maximum threshold duration and to a minimum threshold duration; and

generating a system alarm if more than a predetermined number of the last N incoming calls fall below the minimum threshold duration or exceed the maximum threshold duration.

39. The method as set forth in claim 38 wherein the step of generating a system alarm comprises the step of generating a short call alarm if more than three of the last ten incoming calls fall below the minimum threshold duration.

40. The method as set forth in claim 38 wherein the step of generating a system alarm comprises the step of generating a long call alarm if more than three of the last ten incoming calls exceed the maximum threshold duration.

41. The method as set forth in claim 38 wherein the step of generating a system alarm comprises the step of generating a long call alarm if two consecutive incoming calls exceed the maximum threshold duration.

42. The method as set forth in claim 38 including the further step of sending the system alarm to a computer terminal coupled to the automated telephone system.

43. The method as set forth in claim 38 including the further step of sending the system alarm to a remote station by causing the automated telephone system to initiate an outbound telephone call to the remote station.

44. A method of diagnosing the operation of an automated telephone system coupled to a plurality of telephone lines, wherein incoming call are received in a sequential distribution across the plurality of telephone lines, the method comprising the steps of:

monitoring the telephone lines to detect an initiation and a termination of each incoming call on each telephone line;

measuring a duration of each incoming call;

counting a total number of incoming calls received on each telephone line on a line-by-line basis;

calculating a system call duration average for all incoming calls;

calculating a system call arrival rate for all incoming calls;

calculating an expected call distribution of the incoming calls on each of the telephone lines based on the system call duration average and the system call arrival rate;

comparing a first expected call distribution of a first selected telephone line to an actual call distribution of incoming calls on the first selected telephone line; and

generating a system alarm corresponding to the first selected telephone line if a difference between the first expected call distribution and the actual call distribution exceeds a maximum value.

45. The method as set forth in claim 44 wherein at least one of the values calculated is a weighted average.

46. The method as set forth in claim 44 including the further step of sending the system alarm to a computer terminal coupled to the automated telephone system.

47. The method as set forth in claim 44 including the further step of sending the system alarm to a remote station by causing the automated telephone system to initiate an outbound telephone call to the remote station.

48. An apparatus for diagnosing the operation of an automated telephone system coupled to a plurality of telephone lines, said apparatus comprising:

line monitoring apparatus for detecting an initiation and a termination of each call on each of said telephone lines coupled to said automated telephone system, measuring a duration of each said call, and counting a total number of calls on a line-by-line basis; and

a processor for calculating a first selected statistical parameter for a selected one of said telephone lines;

means for sending said calculated first selected statistical parameter to a monitoring point operating in common with at least two other automated telephone systems, said monitoring point operative to receive like calcu-

lated selected statistical parameters from such other systems;

means for determining when a sent calculated statistical parameter from said system statistically varies from the combined like sent statistical parameters of said other systems; and

means controlled by said determining means when a said variation is deleted for establishing an alarm condition.

49. The invention set forth in claim 48 wherein said statistical parameter is a measure of line utilization for incoming calls.

50. The invention set forth in claim 48 wherein said statistical parameter is a measure of call duration on each call.

* * * * *

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United States Patent [19]

Javitt

[11] **Patent Number: 5,787,162**

[45] **Date of Patent: Jul. 28, 1998**

[54] **APPARATUS AND METHOD FOR SCHEDULING URGENT COMMUNICATION SESSIONS**

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[73] Assignee: **AT&T Corp.**, Middletown, N.J.

[21] Appl. No.: **529,704**

[22] Filed: **Sep. 18, 1995**

[51] **Int. Cl.**^{6} **H04M 3/42**

[52] **U.S. Cl.** **379/229**; 379/201; 379/202; 379/387; 370/259; 370/260; 370/270

[58] **Field of Search** 379/201, 202, 379/211, 212, 58, 61, 207, 265, 266, 57, 210, 243, 229, 230, 231, 387; 370/260, 270, 259

[56] **References Cited**

U.S. PATENT DOCUMENTS

4,313,035	1/1982	Jordan et al.	379/207
4,796,293	1/1989	Blinken et al.	379/202
4,893,335	1/1990	Fuller et al.	379/200
4,910,766	3/1990	Ogino et al.	379/243
5,153,905	10/1992	Bergeron et al.	379/209
5,243,645	9/1993	Bissell et al.	379/211
5,278,898	1/1994	Cambray et al.	379/266
5,311,583	5/1994	Friedes et al.	379/209
5,327,486	7/1994	Wolff et al.	379/211
5,329,578	7/1994	Brennan et al.	379/67
5,349,649	9/1994	Iijima	395/275
5,384,831	1/1995	Creswell et al.	379/211
5,386,512	1/1995	Crisman et al.	395/200
5,434,908	7/1995	Klein	379/88
5,434,984	7/1995	Delodddere et al.	395/288
5,500,889	3/1996	Baker et al.	379/243
5,546,449	8/1996	Hogan et al.	379/202
5,548,636	8/1996	Bannister et al.	379/210
5,610,970	3/1997	Fuller et al.	379/210
5,625,680	4/1997	Foladare et al.	379/243

FOREIGN PATENT DOCUMENTS

WO 91/07838	4/1991	WIPO	H04M 11/00

Primary Examiner—Krista Zele
Assistant Examiner—Scott Wolinsky
Attorney, Agent, or Firm—Alfred G. Steinmetz

[57] **ABSTRACT**

Communication systems, terminals and methods are provided which are capable of supporting, and/or being used in conjunction with, communication session scheduling. The communication systems include a plurality of terminals, ones of which are operative to receive and transmit communication signals among a plurality of parties. At least one terminal includes both an interface operative to receive a scheduling signal and a circuit. The circuit is operative in response to the received scheduling signal to modify selectively one of either a first data set or a second data set. The first data set represents the availability of a first party to respond to a received communication signal. The second data set represents the priority associated with a particular communication signal transmitted from the first party to at least a second party.

30 Claims, 6 Drawing Sheets

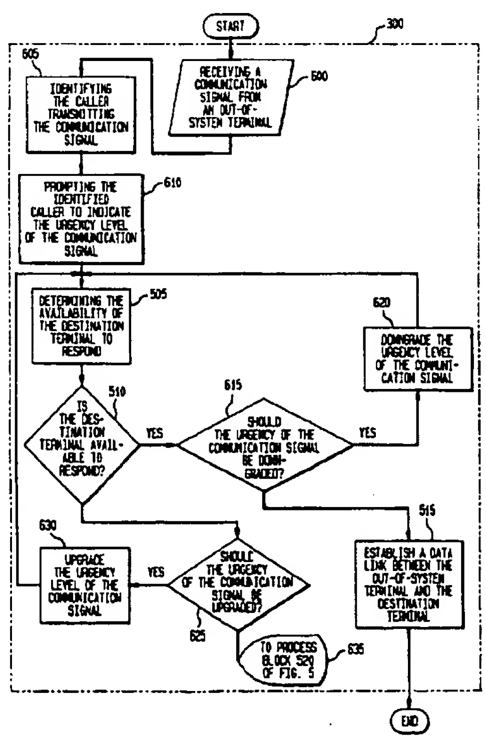


FIG. 1A

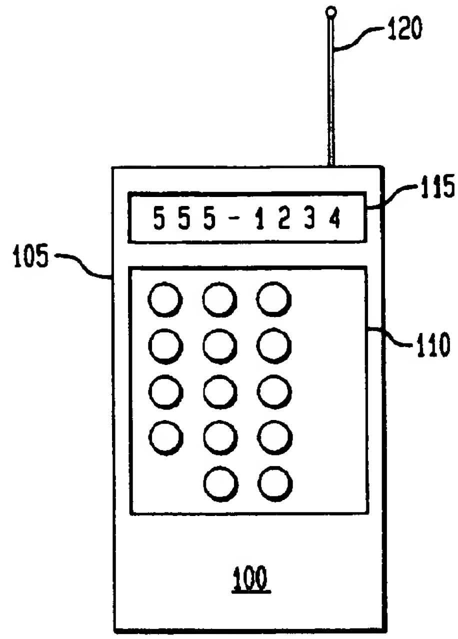

120
115
555-1234
105
110
100

FIG. 1B

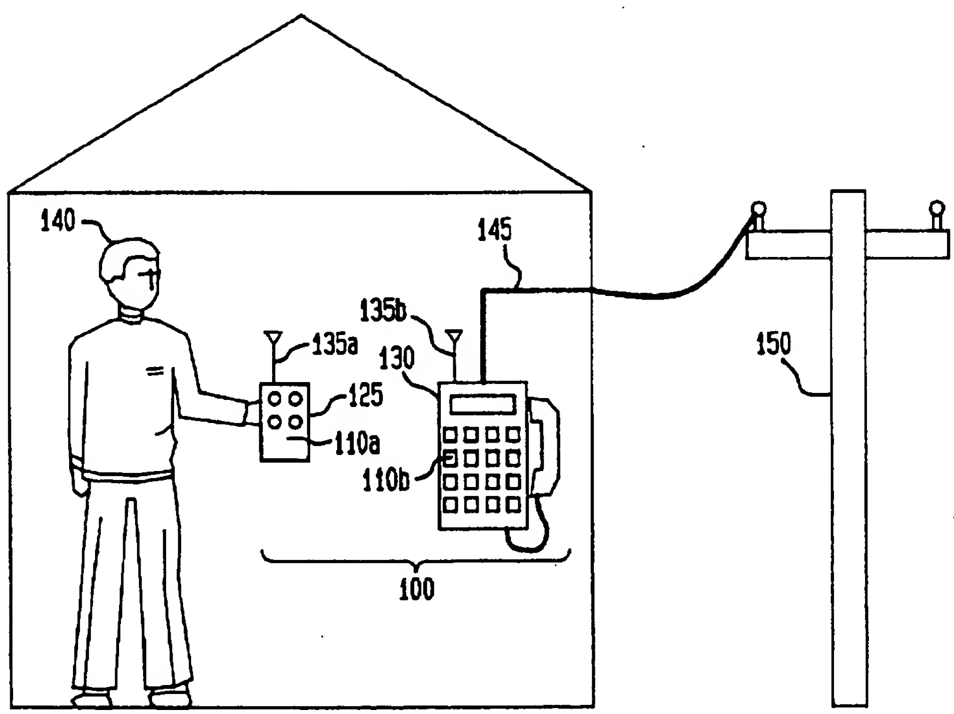

140
145
135b
135a
130
125
110a
110b
150
100

FIG. 2

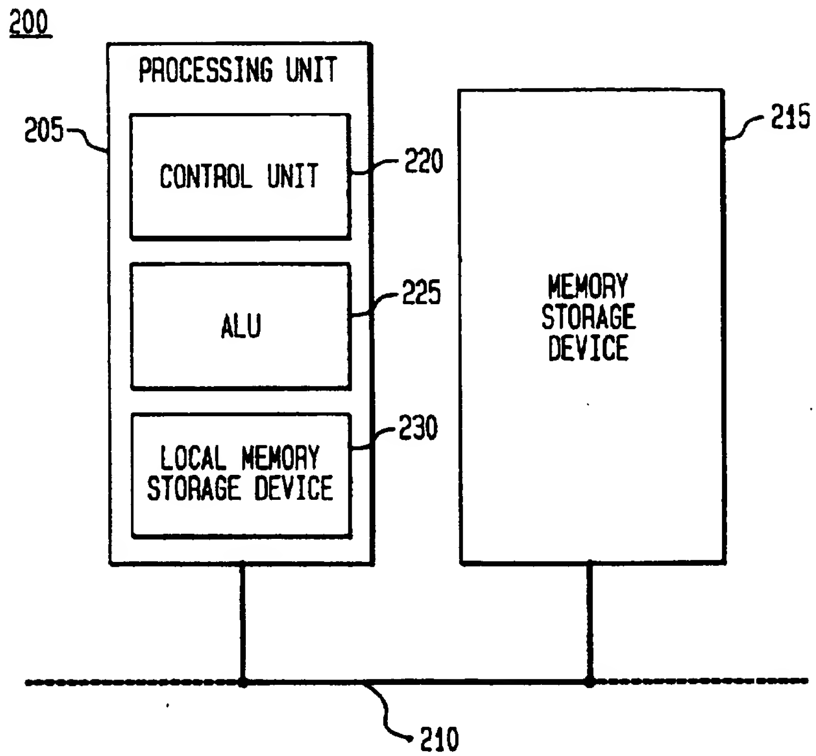
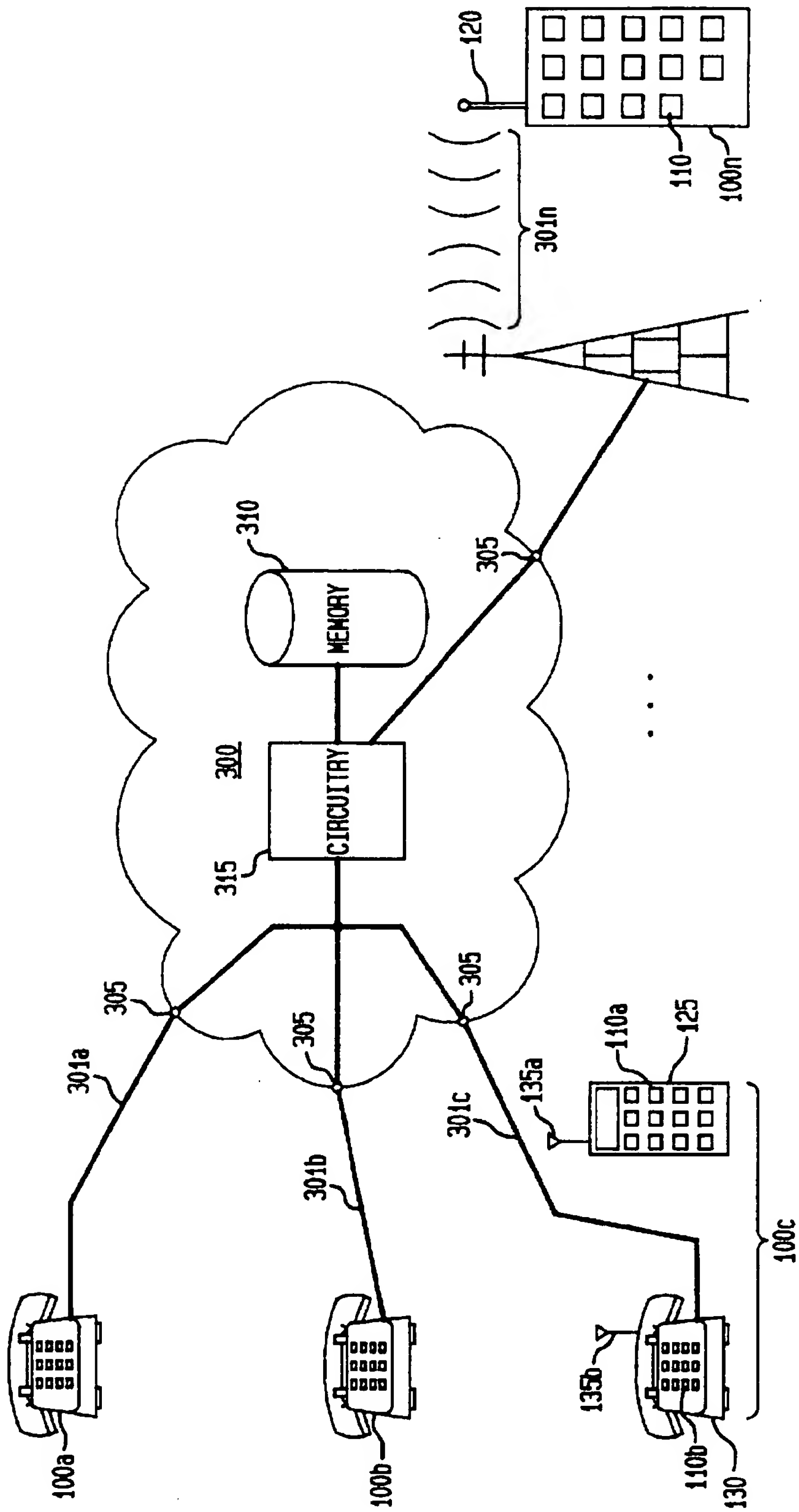
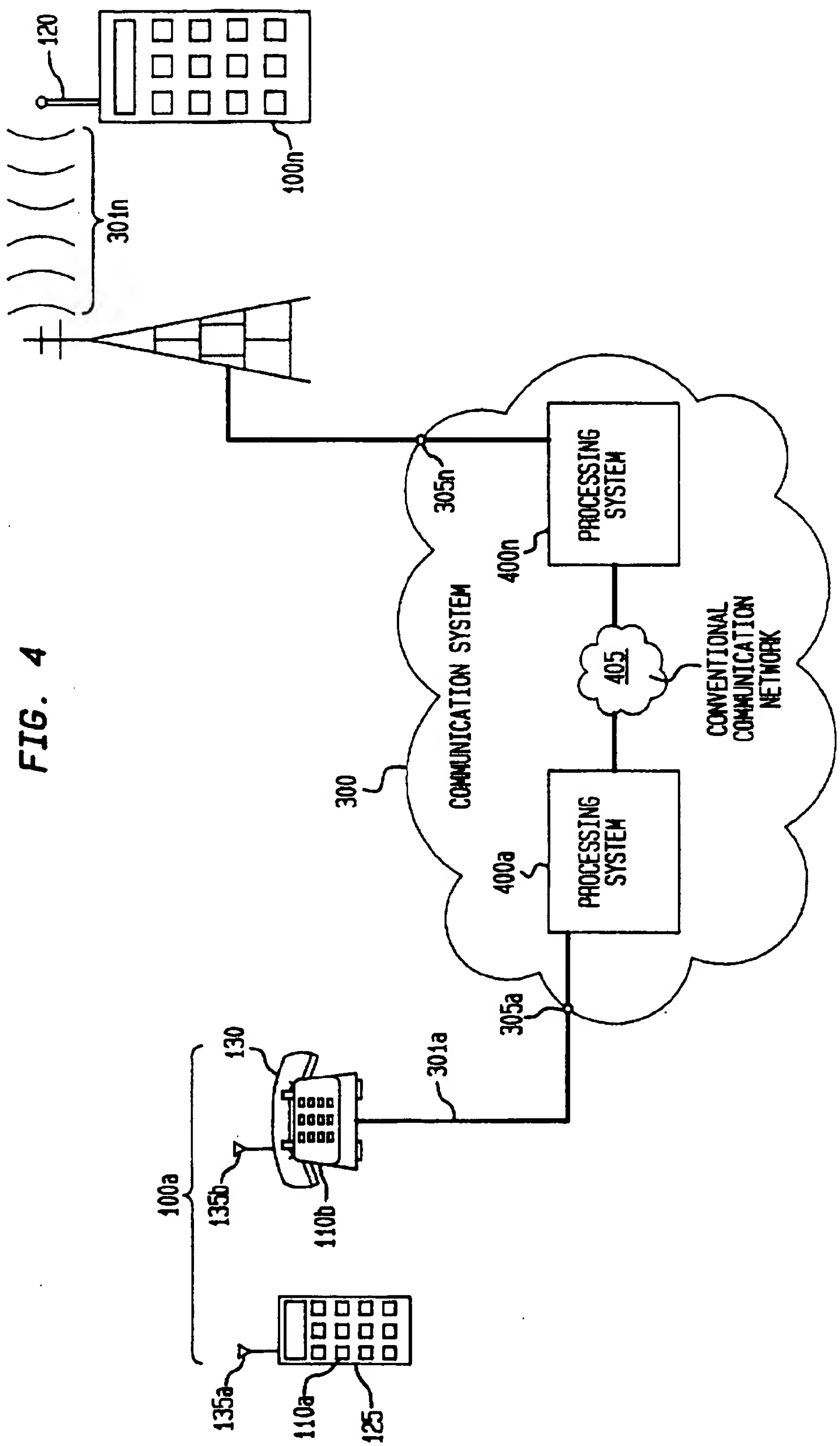

200
PROCESSING UNIT
205
CONTROL UNIT
220
ALU
225
LOCAL MEMORY STORAGE DEVICE
230
MEMORY STORAGE DEVICE
215
210

FIG. 3


120
110
100n
301n
310
MEMORY
305
300
CIRCUITRY
315
305
305
305
301a
301b
301c
135a
110a
125
100c
100a
100b
135b
110b
130



120
100n
301n
305n
PROCESSING SYSTEM
400n
COMMUNICATION SYSTEM
405
CONVENTIONAL COMMUNICATION NETWORK
300
PROCESSING SYSTEM
400a
305a
301a
130
100a
135b
110b
135a
110a
125

FIG. 4

FIG. 5

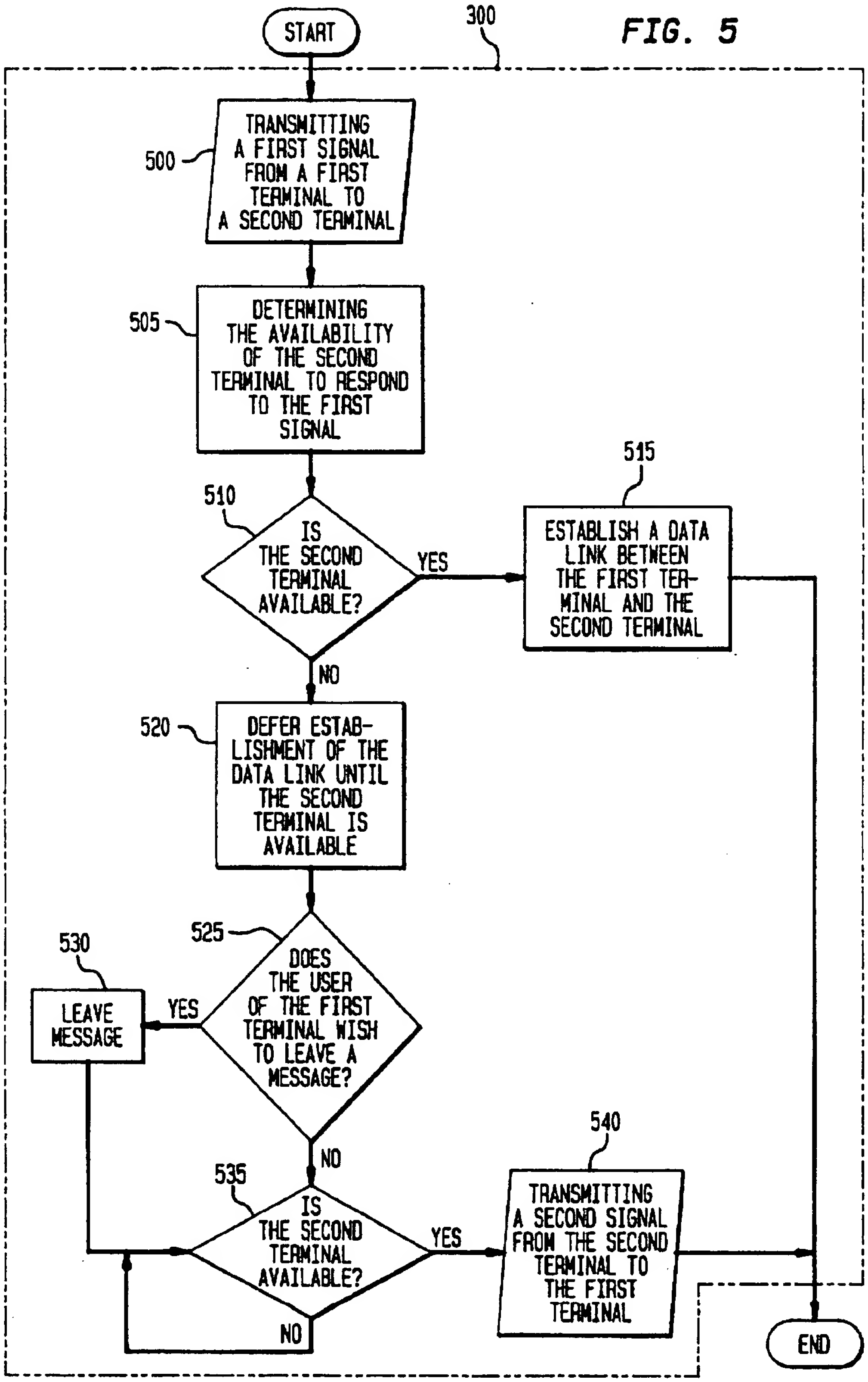
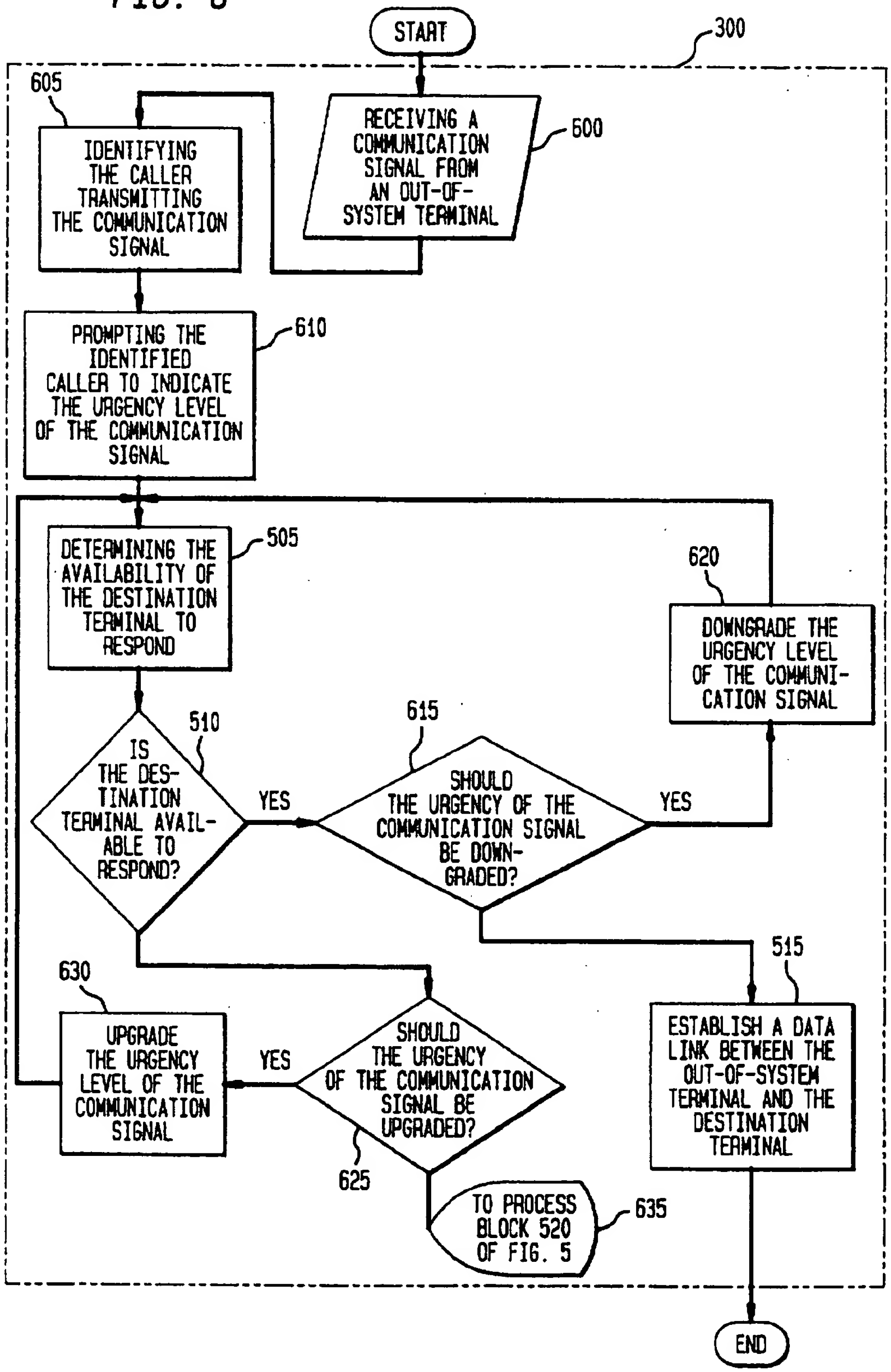

START
300
TRANSMITTING A FIRST SIGNAL FROM A FIRST TERMINAL TO A SECOND TERMINAL
500
DETERMINING THE AVAILABILITY OF THE SECOND TERMINAL TO RESPOND TO THE FIRST SIGNAL
505
510
IS THE SECOND TERMINAL AVAILABLE?
YES
NO
515
ESTABLISH A DATA LINK BETWEEN THE FIRST TERMINAL AND THE SECOND TERMINAL
520
DEFER ESTABLISHMENT OF THE DATA LINK UNTIL THE SECOND TERMINAL IS AVAILABLE
525
DOES THE USER OF THE FIRST TERMINAL WISH TO LEAVE A MESSAGE?
YES
NO
530
LEAVE MESSAGE
535
IS THE SECOND TERMINAL AVAILABLE?
YES
NO
540
TRANSMITTING A SECOND SIGNAL FROM THE SECOND TERMINAL TO THE FIRST TERMINAL
END

FIG. 6



START
300
605
IDENTIFYING THE CALLER TRANSMITTING THE COMMUNICATION SIGNAL
RECEIVING A COMMUNICATION SIGNAL FROM AN OUT-OF-SYSTEM TERMINAL
600
PROMPTING THE IDENTIFIED CALLER TO INDICATE THE URGENCY LEVEL OF THE COMMUNICATION SIGNAL
610
DETERMINING THE AVAILABILITY OF THE DESTINATION TERMINAL TO RESPOND
505
620
DOWNGRADE THE URGENCY LEVEL OF THE COMMUNICATION SIGNAL
510
IS THE DESTINATION TERMINAL AVAILABLE TO RESPOND?
YES
615
SHOULD THE URGENCY OF THE COMMUNICATION SIGNAL BE DOWNGRADED?
YES
515
630
UPGRADE THE URGENCY LEVEL OF THE COMMUNICATION SIGNAL
YES
SHOULD THE URGENCY OF THE COMMUNICATION SIGNAL BE UPGRADED?
625
ESTABLISH A DATA LINK BETWEEN THE OUT-OF-SYSTEM TERMINAL AND THE DESTINATION TERMINAL
TO PROCESS BLOCK 520 OF FIG. 5
635
END

APPARATUS AND METHOD FOR SCHEDULING URGENT COMMUNICATION SESSIONS

TECHNICAL FIELD OF THE INVENTION

The present invention relates generally to the field of telephony, and more particularly, to terminals, methods and systems which provide session scheduling for intelligent communication.

BACKGROUND OF THE INVENTION

Telecommunications involves the transmission of speech and data between two or more points using electric signals. Representative communication media include air, wire, fiber optic cable, and the like.

Through conventional communication systems and emerging technologies, including without limitation wired, wireless and video telephony, pagers, e-mail and voice mail systems, and the like, individual "reachability" has significantly increased. In so doing, modern telecommunications has failed to provide for convenient direct person to person communication.

Business people, for example, are often required to leave multiple telephone numbers where, and/or beeper numbers through which, they may be reached. Callers are required to understand the different communication technologies available to reach a particular party and to determine the optimum approach, considering cost, quality, availability, etc., to reach that party at a given time.

Conventional approaches have also failed to address problems associated with information overload and time management. Called parties are often interrupted by an incoming call causing a loss of productivity while performing some activity. Worse, the party called often decides that the interrupting call is less important than the present activity and a response to the same may wait until a more appropriate time.

Alternatively, two individuals may wish to have a conversation but neither is available at the same time. Both parties waste time and effort receiving messages to call the other back, only to find that the other party is not available. This phenomenon is commonly referred to as "telephone tag". This problem is compounded further when three or more parties wish to have a conference call.

The effects of the foregoing deficiencies remain a dominant obstacle to producing more efficient, intelligent and commercially desirable telephony based systems and products.

SUMMARY OF THE INVENTION

Broadly, the present invention is directed to terminals, methods and systems which provide session scheduling for intelligent communication. The invention provides communication management functionality that requires only a minimum of user control. Conversations between parties are enabled automatically and in an optimum manner avoiding unnecessary and annoying interruptions. A calling party simply specifies the party or parties with whom he wishes to speak and the urgency of the call. The parties are automatically connected as soon as they are available for a call of the stated priority.

The principles of the present invention are particularly beneficial when utilized to provide a data link for communication with wired, wireless (e.g., vehicular and hand held cellular telephones) and video telephony, hand-held/

personal/notebook/laptop computers ("PCs"), personal communication assistants ("PCAs"), suitably arranged pay-telephones, message paging, e-mail, voice mail and the like.

An exemplary terminal in accordance with the principles of the present invention may suitably be used with a communication system capable of supporting communication session scheduling. The terminal includes receiving and transmitting means, an interface and a circuit.

The receiving and transmitting means are operative to receive and transmit communication signals between a first party and at least a second party via the communication system. The interface is operative to receive a scheduling signal. The circuit is operative in response to the received scheduling signal to modify selectively at least one of either a first data set or a second data set. The first data set represents the availability of the first party to respond to a particular received communication signal. The second data set represents the urgency associated with a particular communication signal transmitted from the first party to at least the second party.

An important aspect of the invention is that the scheduling signal may suitably be received from either the first party or a scheduling apparatus.

A system in accordance with the principles of the present invention operates to provide selectively one or more data links between ones of a plurality of terminals. Each one of the terminals is operative to transmit and receive one or more signals via the one or more selectively provided data links. An exemplary system includes an input port, a memory and circuitry.

The input port is operative to receive a first signal transmitted from a first terminal to a second terminal. A portion of the first signal represents the urgency associated with the first signal. The memory is operative to store a data set representing the availability of the second terminal to respond to the first signal. The circuitry is operative to compare the data set and the portion of the first signal. In response to the comparison, the circuitry may provide a data link between the first terminal and the second terminal. Otherwise, the circuitry will transmit, as a function of the data set and the portion of the first signal, a second signal from the second terminal to at least the first terminal. At least a portion of the second signal represents the urgency associated with the second signal.

Another important aspect is that any system, terminal and, in particular, any circuitry, in accordance with the principles of the present invention may suitably be firmware, hardware or processing system based.

A method in accordance with the principles of the present invention concerns selectively scheduling communication sessions between ones of a plurality of terminals in a communication system. Each one of the terminals is operative to transmit and receive one or more signals via the communication system. A first signal is transmitted from a first terminal to a second terminal. A portion of the first signal represents the urgency of a communication session between the first terminal and the second terminal. The availability of the second terminal to respond to the first signal is determined using at least the portion of the first signal. The communication session is established upon a determination that the second terminal is available, otherwise establishment of the communication session is deferred until the second terminal is available, preferably as a function of the portion of the first signal.

One embodiment for using and/or distributing the present invention is as software. The software embodiment includes

a plurality of processing instructions which are stored to a conventional storage medium. The instructions are readable and executable by one or more processing systems having one or more processing units. The instructions, upon execution, control the one or more processing units to provide communication session scheduling in a communication system. Preferred storage media include without limitation magnetic, optical, and semiconductor, as well as suitably arranged combinations thereof.

The foregoing has outlined, rather broadly, preferred and alternative features of the present invention so that those skilled in the art may better understand the detailed description of the invention that follows. Additional features of the invention will be described hereinafter that form the subject of the claims of the invention. Those skilled in the art should appreciate that they can readily use the disclosed conception and specific embodiment as a basis for designing or modifying other structures for carrying out the same purposes of the present invention. Those skilled in the art should also realize that such equivalent constructions do not depart from the spirit and scope of the invention in its broadest form.

BRIEF DESCRIPTION OF THE DRAWINGS

For a more complete understanding of the present invention, reference is now made to the following descriptions taken in conjunction with the accompanying drawings in which like designations represent like parts, and in which:

FIG. 1A illustrates an exemplary wireless terminal in accordance with the principles of the present invention;

FIG. 1B illustrates an exemplary terminal having at least two apparatus in accordance with the principles of the present invention;

FIG. 2 illustrates a block diagram of one exemplary microprocessing system which may suitably be utilized in conjunction with the exemplary terminals of FIGS. 1A and 1B;

FIG. 3 illustrates a block diagram of an exemplary communication system capable of supporting communication session scheduling in accordance with the principles of the present invention;

FIG. 4 illustrates a block diagram of another exemplary communication system in accordance with the principles of the present invention;

FIG. 5 illustrates a flow diagram for selectively scheduling communication sessions between ones of a plurality of terminals of a communication system in accordance with the principles of the present invention; and

FIG. 6 illustrates another flow diagram for selectively scheduling communication sessions in accordance with the principles of the present invention.

DETAILED DESCRIPTION

FIGS. 1A and 1B illustrate exemplary terminals that may suitably be arranged to operate in accordance with the principles of the present invention. Each of the terminals is operative in conjunction with a suitably arranged communication system that is capable of supporting communication session scheduling. "Communication session scheduling" as used herein includes without limitation establishing a data link or connection at an optimum time between two or more terminals enabling communication therebetween. "Data link" as used herein includes without limitation any suitable hardware, firmware and software configuration that enables a plurality of terminals to communicate, i.e., to transmit and receive one or more signals or data packets, such as over a communication channel. "Terminal" as used herein includes without limitation any station, device, apparatus and/or the like which is used to transmit and receive one or more signals. "Signal" as used herein includes without limitation data packets, sequence of data, or any other variation of a physical quantity used to convey information. Exemplary communication systems will be discussed in greater detail with reference to FIGS. 3 to 6.

FIG. 1A illustrates an exemplary terminal **100**, a wireless telephone, having a housing **105**, a keypad **110**, a display **115** and at least one suitably arranged antenna **120**. Keypad **110** preferably includes a set of keys or control levers having a systematic arrangement that is used to operate wireless telephone **100**. Display **115** provides a visible representation of information, such as a telephone number for example, presented in graphical or character form. Wireless antenna **120** is operative to receive and transmit communication signals between a first party using wireless telephone **100** and at least a second party via a suitable communication system.

Keypad **110** is also operative to provide an interface through which one or more scheduling signals may be received from a user of wireless telephone **100**. "Scheduling signals" as used herein includes without limitation electromagnetic signals used to select an urgency level for a particular call a user is making and/or a priority level based on the importance of the user's current activity as compared with the urgency of any incoming call.

One example of a suitable priority and/or urgency coding scheme would use "1" for extreme emergency to "9" for a low level activity, with values therebetween representing various levels of activity. The development of an appropriate implementation/scheme of scheduling signals is applications specific. As will be discussed with reference to FIGS. 3 to 6, the user will only be alerted as to the reception of an incoming communication if the urgency of the communication is equal to or exceeds the user's current priority level.

In alternate embodiments, scheduling signals may suitably be received from any other suitably arranged input device or alternatively a scheduling apparatus, such as a PC, PDA or the like, running a time management program. Such scheduling signals may suitably be received over a wired or wireless data link through a parallel or serial port (not shown) of wireless telephone **100**.

FIG. 1B illustrates another exemplary terminal **100** in accordance with the principles of the present invention. Exemplary terminal **100** includes at least two exemplary apparatus, or devices, wherein a first apparatus is a portable communication device **125** and a second is a wired telephone **130**. Each exemplary apparatus **125**, **130** includes an antenna **135a**, **135b** and a keypad **110a**, **110b**, respectively. Each antenna **135a**, **135b** is suitably arranged to provide a data link for wireless communication between first and second apparatus **125**, **130**, respectively. Each keypad **110a**, **110b** preferably includes a set of keys or control levers, again having a systematic arrangement, used to operate portable communication device **125** and wired telephone **130**, respectively.

Wired telephone **130** is operative to receive and transmit communication signals between a user **140** and at least another party. The foregoing is accomplished using a suitable communication system **145**, **150** to which wired telephone **130** is coupled.

In the illustrated embodiment, keypad **110a** is further operative to provide an interface through which one or more scheduling signals may suitably be received. Again, in

alternate embodiments, scheduling signals may be received from another suitably arranged input device or alternatively a suitable scheduling apparatus running a time management program.

An aspect of any multiple apparatus terminal in accordance with the present invention is that first apparatus **125** may suitably be implemented in a compact and light embodiment capable of being continuously worn by a user. Further, second apparatus **130** may in point of fact be wired, as shown, or wireless.

Other preferred embodiments, whether embodied in a single or multiple apparatus terminal, include without limitation vehicular wireless telephones, suitably arranged PCs and PDAs, specially equipped pay telephones, and the like. Another important aspect is that the receiving and transmitting means may suitably be configured to receive and/or transmit either voice and/or data communication signals. Further, terminal **100** may suitably include means for indicating that a communication signal has been received, including without limitation audio, visual, physical sensation or the like.

It should be noted that the interface provided by keypad **110** and **110***a* of FIGS. 1A and 1B, respectively, are illustrative only and that the associated functionality may suitably be provided by any means operative to receive a scheduling input and in response thereto to generate a scheduling signal, including without limitation speech or hand writing recognition.

In addition to the services which will be described with reference to FIGS. **3** to **6**, terminal **100** may suitably be arranged to provide other communication related features, including without limitation, world wide direct access codes, password protected credit card numbers, time zone conversion and the like. Terminal **100** may also include a personal directory and function as an auto-dialer through data link or Dual-Tone Multi-Frequency ("DTMF") generation.

FIG. **2** illustrates a block diagram of one exemplary microprocessing system **200** which may suitably be utilized in conjunction with the exemplary terminals of FIGS. 1A and 1B. Microprocessing system **200** includes a single processing unit **205** coupled via data bus **210** with a single memory storage device **215**. Memory storage device **215** is suitably operative to store data and/or one or more processing system instructions which processing unit **205** is operative to retrieve and execute. Memory storage device **215** may be any suitable memory storage device or plurality thereof.

Processing unit **205** includes a control unit **220**, an arithmetic logic unit ("ALU") **225** and a local memory storage device **230** (e.g., stackable cache, a plurality of registers, etc.). Control unit **220** is operative to fetch processing system instructions from memory storage device **215**. ALU **225** is operative to perform a plurality of operations, including addition and Boolean AND, needed to carry out those instructions. Local memory storage device **230** is operative to provide local high speed storage used for storing temporary results and control information.

In accordance with the illustrated embodiment, microprocessing system **200** is operative to receive, possibly via data bus **210**, a scheduling signal. Microprocessing system **200** operates, in response to the received scheduling signal, to modify selectively at least one of a first data set and a second data set. Either or both of the data sets are preferably stored in at least one of either memory storage device **215** or local memory storage device **230**. The first data set represents the availability of the first party to respond to a received communication signal, whereas the second data set represents a priority associated with a particular communication signal transmitted from the first party to at least the second party.

In alternate exemplary embodiments, microprocessing system **200** may suitably be replaced by, or combined with, without limitation, programmable logic devices, such as PALs (programmable array logic) and PLAs (programmable logic arrays), DSPs (digital signal processors), FPGAs (field programmable gate arrays), ASICs (application specific integrated circuits), VLSIs (very large scale integrated circuits) and the like. Further, processing unit **205** and memory storage device **215** may be separately implemented in first apparatus **125** and second apparatus **130**, respectively.

FIG. **3** illustrates a block diagram of an exemplary communication system **300** that is capable of supporting communication session scheduling in accordance with the principles of the present invention. System **300** includes a plurality of exemplary terminals **100***a* to **100***n* and is operative to provide selectively one or more data links **301***a* to **301***n* between particular ones of terminals **100***a* to **100***n*. Each one of terminals **100***a* to **100***n* is preferably operative to transmit and receive one or more signals via the one or more selectively provided data links **301***a* to **301***n*.

System **300** includes without limitation at least one input port **305**, at least one memory **310** and circuitry **315**. Input port **305** is operative to receive a first signal transmitted from a first terminal, such as terminal **100***b*, to at least a second terminal, such as terminal **100***n*, for example. At least a portion of the first signal represents an urgency level associated with the first signal. As discussed with reference to FIGS. 1A and 1B, interface **110** enables a user to select or specify the urgency level when transmitting a communication. The urgency level indicates the speed with which the first signal must be responded to by second terminal **100***n*.

A memory associated with each terminal, such as memory storage device **215** or memory **310**, for example, is operative to store at least one data set, which represents the availability of second terminal **100***n* to respond to the first signal. Also as discussed with reference to FIGS. 1A and 1B, interface **110** enables user **140** to select or specify a current priority level based on the importance of its current activity as compared to the particular urgency associated with one or more incoming calls.

Circuitry **315** is operative to compare the data set and the portion of the first signal representing the urgency factor. If the comparison indicates that second terminal **100***n* is able to respond to the first signal, then circuitry **315** is operative to provide a data link between first terminal **100***b* and second terminal **100***n*. Second terminal **100***n* alerts user **140** to reception of the first signal only if the urgency associated therewith equals or exceeds second terminal **100***n*'s current availability or priority level. Establishment of a data link allows terminals **100***b* and **100***n* to suitably transmit voice and/or data signals, such as video data, therebetween.

If the comparison indicates that second terminal **100***n* is unavailable to respond to the first signal, then circuitry **315** is operative to transmit, as a function of the data set and the portion of the first signal, a second signal from second terminal **100***n* to at least first terminal **100***b*. At least a portion of the second signal represents the urgency associated with the second signal, i.e., the urgency with which the first signal must be responded to by first terminal **100***b*. Recall, that the development of a preferable urgency and/or availability coding scheme is application dependent.

FIG. **4** illustrates a block diagram of another exemplary communication system **300** in accordance with the prin-

ciples of the present invention. Communication system 300 includes a plurality of processing systems 400a to 400n which are in operative communication with one another via a conventional communication network 405. Each processing system 400 preferably includes one or more processing units in operative communication with one or more memory storage devices.

It should be noted that any processing system capable of functioning in accordance with communication system 300 and/or communication network 405 may suitably replace, or be utilized in conjunction with, any of processing systems 400, including without limitation, videophones, telephones, televisions, sophisticated calculators and, hand-held, laptop/notebook, personal, mini, mainframe and super computers, including RISC and parallel processing architectures, as well as within processing system network combinations of the foregoing. Conventional processing system architecture is more fully discussed in *Computer Organization and Architecture*, by William Stallings, MacMillan Publishing Co. (3rd ed. 1993), which is incorporated herein by reference. Alternatively, any of processing systems 400 may suitably be combined with or replaced by, without limitation, DSPs, FPGAs, ASICs, VLSIs, and the like, in accordance with the present invention as previously discussed.

It should be noted also that although each terminal 100 is shown suitably coupled with a single processing system 400, in point of fact, each terminal 100 may suitably be coupled with a plurality of processing systems 400. Similarly, each processing system 400 may suitably be coupled with a plurality of terminals 100, such as a Public Switched Telephone Network ("PSTN") switching office for example.

FIG. 5 illustrates a flow diagram for selectively scheduling communication sessions between ones of a plurality of terminals 100 of communication system 300 in accordance with the principles of the present invention. Each terminal 100 is preferably operative to transmit and receive one or more signals via communication system 300.

A first signal is initiated and transmitted from a first terminal to at least a second terminal (input/output block 500). At least a portion of the first signal preferably represents an urgency level, i.e., the urgency with which a data link or communication session must be established between the first and second terminals.

In other words, a user initiates a communication session by indicating one or more parties to call and associating an urgency level with the communication session and/or each particular call. In addition, the caller may enter a deadline for the communication session or each call, an estimated length for the communication session, a topic statement or the like. The user's terminal 100 accesses communication system 300 which is suitably operative to transmit a calling signal to each of the called parties.

Communication system 300 is preferably operative to identify an incoming caller using conventional techniques, such as retrieving relevant information from memory if the caller is a subscriber of the same ISDN PBX system, for example, or alternatively, to use conventional ANI techniques to identify the caller. System 300 may also prompt the caller for other information.

System 300 operates to determine the availability of the second terminal to respond to the first signal using that portion of the first signal representing its urgency level (processing block 505). System 300 also preferably accesses and uses a data set from memory, the data set representing the availability of second terminal to respond to the first signal.

Referring to FIG. 4 for illustration, processing system 400n is operative to receive the calling signal from processing system 400a and if second terminal 100n is available (YES branch of decision block 510), processing system 400n is operative to establish a data link, or communication session, between terminals 100a and 100n(process block 515). In other words, the party using second terminal 100n is alerted with relevant information regarding the calling party.

In the event that second terminal 100n is unavailable, processing system 400n is operative to defer establishment of the data link, or communication session, until second terminal 100n becomes available. This is preferably accomplished as a function of at least the portion of the calling signal representing the urgency of the call (process block 520).

When deferring the call, second terminal 100, through processing system 400n, may suitably reduce the calling signal's urgency and/or increase the priority of its own current activity. If the user of first terminal 100a desires to leave a message for the user of second terminal 100n(YES branch of decision block 525), System 300 is operative to connect the user with a voice and/or data recording system, such as voice mail, e-mail, paging services or the like (process block 530).

If the calling signal does not include a data set or subset indicating whether the call must be returned by a specific deadline, system 300 preferably prompts the caller to indicate same. For example, the caller may have indicated the call was "significant" (priority level equals 6) but it may be "critical" (priority level equals 3) that the parties speak before 5 p.m. tomorrow. Such information may be received from the caller using an automated system with Dual-Tone Multi-Frequency ("DTMF") or speech recognition, for example. System 300 preferably uses the received deadline and estimated length information, and possibly other information, to schedule a convenient return communication session request.

System 300 is operative to monitor second terminal 100n to identify any change in availability to respond to the communication session request of first terminal 100a (decision block 535). For example, following a meeting, the user of second terminal 100n may return to his office and modify his priority level, lowering the same to reflect the importance of his current activity. Processing system 400n will begin the process of completing deferred calls having an urgency level greater than or equal to the current priority level setting.

Thus, when second terminal 100n becomes available (YES branch of decision block 535), processing system 300n is operative to transmit, as a function of the portion of the first signal and second terminal 100n's availability, a second signal from second terminal 100n to at least first terminal 100a(input/output block 540). At least a portion of the second signal represents the urgency associated therewith.

Thus processing system 400n will attempt to communicate with processing system 400a to establish a communication session between first and second terminals 100a and 100n, respectively. This activity is invisible to the users of either terminal until both are suitably available for the communication session. If first terminal 100a is available, the user of second terminal 100n is alerted. Preferably, the name of the user of first terminal 100a, the urgency associated with the call, the call's topic and estimated duration, an indication that the call was deferred, and/or the like is displayed.

In the event that first terminal 100a is unavailable, processing system 400a will defer establishment of the communication session. An important aspect of the invention is the substantial, if not complete, elimination of "telephone tag". System 300 is operative to automatically connect a plurality of terminals at a time that is mutually convenient.

In particular embodiments, upon becoming available, the user may be offered an option to review, and possibly reprioritize, deferred calls either by modifying their assigned urgency level and/or by directly overriding the order in which they are awaiting response, for example. In other embodiments, deferred calls may be responded to according to priority, associated deadlines, estimated call length, period of time that the call has remained deferred, and/or the like. In still further embodiments, if the user's schedule has been received from a suitably arranged scheduling apparatus, such as a PC, PDA or the like, running a time management program, the scheduling apparatus may anticipate the date, time and priority of the user's next activity and not return a particular deferred call having an estimated call length that is greater than the time remaining before the user's next activity.

FIG. 6 illustrates a flow diagram for processing communication signals received from a first time caller, or possibly a conventional terminal inoperative to support communication session scheduling in accordance with the present invention. Although implementation of the present invention is envisioned as servicing large work groups so that most communication would be between system subscribers, system 300 may suitably offer assistance when completing deferred calls to non-system subscribers. System 300 would still prioritize the calls. Further, the subscriber may select to have system 300 call with a synthesized voice and alert the subscriber when the other party picks up or leaves a message.

More particularly, a communication signal is received by communication system 300 from a first time caller or an out-of-system terminal (input/output block 600). For illustrative purposes, FIG. 6 refers to a call received from an out-of-system terminal. The caller transmitting the signal is identified as discussed previously (process block 605). The identified caller is prompted and asked to indicate the urgency level to be associated with the call (process block 610).

Communication system 300 operates to determine the availability of the destination terminal to respond to the same (process block 505). The determination is made using the indicated urgency level and, preferably, the current priority level of the destination terminal.

If the destination terminal is available, then a data file may be accessed to determine whether the urgency level of the communication signal should be downgraded (decision block 615). Communication system 300 may suitably be programmed to automatically decrease the priority level associated with a particular call, such as from a salesperson that has consistently mis-characterized the urgency of his calls (process block 620). If the communication signal is not downgraded then a data link is established between the first time call or out-of-system terminal and the destination terminal (process block 515).

In the event that the destination terminal is unavailable, another data file may be accessed to determine whether the urgency level of the communication signal should be upgraded (decision block 625). Communication system 300 may suitably be programmed to automatically increase the priority level associated with a particular call, such as from an important customer (process block 630). If the communication signal is not upgraded then establishment of a data link between the out-of-system terminal and the destination terminal is deferred, as discussed with reference to FIG. 5 (connector block 635).

Importantly, communication system services in accordance with the present invention may suitably be "personalized" for particular individuals. For example, individuals who are generally in their offices and do not require or wish to use pager and/or full personal communication system services may receive interruption management, messaging, and/or deferred call management services only.

Although the present invention has been described in detail, those skilled in the art should understand that they can make various changes, substitutions and alterations herein without departing from the spirit and scope of the invention in its broadest form.

What is claimed is:

1. A terminal for use in a communication system, said communication system capable of supporting communication session scheduling, said terminal comprising:

means for receiving and transmitting communication signals between a first party and a second party via said communication system;

an interface operative to receive an urgency portion of a scheduling signal initiated at the time a call is originated; and

a circuit operative in response to said received urgency portion of a scheduling signal to modify selectively a first data set and a second data set, said first data set representative of the availability of said first party to respond to a received communication signal and said second data set representative of the urgency associated with a particular communication signal transmitted from said first party to said second party.

2. The terminal as set forth in claim 1 further comprising a memory operative to store one or more data sets including said first data set and said second data set.

3. The terminal as set forth in claim 1 wherein at least a portion of said particular communication signal is used to transmit said second data set from said first party to said second party.

4. The terminal as set forth in claim 1 further comprising a first apparatus including said interface and said circuit and a second apparatus including said receiving and transmitting means, said first apparatus and said second apparatus in operative communication with one another.

5. The terminal as set forth in claim 4 wherein said first apparatus is portable and is in operative wireless communication with said second apparatus.

6. The terminal as set forth in claim 1 wherein said receiving and transmitting means is operative to receive at least one of voice communication signals and data communication signals via said communication system.

7. The terminal as set forth in claim 1 further comprising means for indicating reception of a first communication signal.

8. The terminal as set forth in claim 1 wherein said received scheduling signal is received from one of said first party and a scheduling apparatus.

9. A method of operating a terminal of a communication system capable of supporting communication session scheduling, said terminal operative to receive and transmit communication signals between a first party and a second party via said communication system, said method comprising the steps of:

receiving an urgency portion of a scheduling signal initiated at call initialization through an interface of said terminal; and

selectively modifying, in response to receiving said urgency portion of a scheduling signal, a first data set and a second data set,

said first data set representative of the availability of a first party to respond to a received communication signal, and

said second data set representative of the urgency associated with a particular communication signal transmitted from said first party to said second party.

10. The method as set forth in claim 9 further comprising the step of storing in a memory one or more data sets including said first data set and said second data set.

11. The method as set forth in claim 9 further comprising the step of using at least a portion of said particular communication signal to transmit said second data set from said first party to said second party.

12. The method as set forth in claim 9 further comprising the step of indicating that a first communication signal has been received.

13. The method as set forth in claim 12 wherein said first communication signal is at least one of a voice communication signal and a data communication signal.

14. The method as set forth in claim 9 further comprising the step of receiving said scheduling signal from one of said first party and a scheduling apparatus.

15. The method as set forth in claim 9 wherein a plurality of processing instructions are stored to a memory, and said method further comprises the steps of reading and executing one or more of said processing instructions to schedule one or more communication sessions.

16. A circuit for use in a terminal of a communication system capable of supporting communication session scheduling, said circuit comprising:

an input port operative to receive an urgency portion of a scheduling signal initiated at a calling station at initialization of a call,

a memory operative to store a plurality of data sets; and

circuitry operative in response to receiving said urgency portion of a scheduling signal to modify selectively at least one of a first data set and a second data set stored within said memory,

said first data set representative of the availability of said terminal for use in responding to a received communication signal from said communication system, and

said second data set representative of the urgency associated with a communication signal transmitted from said terminal over said communication system.

17. A system for selectively providing data links between ones of a plurality of terminals, each one of said terminals operative to transmit and receive one or more signals via said one or more selectively provided data links, said system comprising:

an input port operative to receive a first signal transmitted from a first terminal to a second terminal, a portion of said first signal created in conjunction with a call origination representing the urgency associated with said first signal:

a memory operative to store a data set representative of the availability of said second terminal to respond to said first signal; and

circuitry operative to compare said data set and said portion of said first signal and, in response thereto, to one of:

provide a data link between said first terminal and said second terminal, and

transmit, as a function of said data set and said portion of said first signal, a second signal from said second terminal to said first terminal, at least a portion of said second signal representing the urgency associated with said second signal.

18. The system as set forth in claim 17 wherein said circuitry is further operative to defer transmitting said second signal until said second terminal is available.

19. The system as set forth in claim 18 wherein transmission of said second signal is deferred as a function of said data set and said portion of said first signal.

20. The system as set forth in claim 18 wherein said circuitry is further operative to adjust at least one of said data set and said portion of said first signal.

21. The system as set forth in claim 18 wherein said circuitry is operative to store a message signal transmitted from said first terminal.

22. The system as set forth in claim 17 wherein said circuitry includes a plurality of processing systems.

23. The system as set forth in claim 22 wherein ones of said processing systems are associated with ones of said plurality of terminals.

24. The system as set forth in claim 17 further comprising a Public Switched Telephone Network (PSTN) switching office.

25. A method for selectively scheduling communication sessions between ones of a plurality of terminals of a communication system, each one of said terminals operative to transmit and receive one or more signals via said communication system, said method comprising the steps of:

transmitting a first signal at call origination from a first terminal to a second terminal, a portion of said first signal created at call origination and representing the urgency of a communication session between said first terminal and said second terminal;

determining the availability of said second terminal to respond to said first signal using at least said portion of said first signal; and

establishing said communication session upon a determination that said second terminal is available, otherwise deferring, as a function of said portion of said first signal, establishment of said communication session until said second terminal is available.

26. The method as set forth in claim 25 wherein said determining step further includes the step of comparing said portion of said first signal and a data set representative of the availability of said second terminal to respond to said first signal.

27. The method as set forth in claim 26 wherein said deferring step further includes the step of transmitting, as a function of said data set and said portion of said first signal, a second signal from said second terminal to at least said first terminal, at least a portion of said second signal representing the urgency associated with said second signal.

28. The method as set forth in claim 27 wherein transmission of said second signal is deferred as a function of said data set and said portion of said first signal.

29. The method as set forth in claim 26 further including the step of adjusting at least one of said data set and said portion of said first signal.

30. The method as set forth in claim 25 further including the step of storing a message signal transmitted from said first terminal.

* * * * *

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United States Patent [19]

Lunn et al.

[11] **Patent Number: 6,075,848**

[45] **Date of Patent: Jun. 13, 2000**

[54] **CALL PATTERNS IN A COMMUNICATIONS NETWORK**

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[73] Assignee: **British Telecommunications public limited company**, London, United Kingdom

[21] Appl. No.: **09/043,295**

[22] PCT Filed: **Sep. 18, 1996**

[86] PCT No.: **PCT/GB96/02331**

§ 371 Date: **Mar. 18, 1998**

§ 102(e) Date: **Mar. 18, 1998**

[87] PCT Pub. No.: **WO97/11547**

PCT Pub. Date: **Mar. 27, 1997**

[30] **Foreign Application Priority Data**

Sep. 18, 1995 [EP] European Pat. Off. 95306560

[51] **Int. Cl.**[7] **H04M 15/00**

[52] **U.S. Cl.** **379/113**; 379/34; 379/133; 379/279

[58] **Field of Search** 379/111–115, 127, 379/133–135, 137–139, 265–266, 309, 279, 34, 120

[56] **References Cited**

U.S. PATENT DOCUMENTS

5,295,183	3/1994	Langlois et al.	379/113
5,592,530	1/1997	Brockman et al.	379/34
5,606,600	2/1997	Elliott et al.	379/112
5,606,601	2/1997	Witzman et al.	379/113
5,854,834	12/1998	Gottlieb et al.	379/113

FOREIGN PATENT DOCUMENTS

2130050	5/1984	United Kingdom .
2204463	11/1988	United Kingdom .
2257869	1/1993	United Kingdom .

OTHER PUBLICATIONS

HET PTT-BEDRIJF, vol. 24, No. 1, Dec. 1886, Den Haag (NL), pp. 30–35, XP000563400 C. Noordegraaf: "Herhaalde Oproepen: Een Studie Naar Abonneegedrag" cited in the application 2; table 1.

Proc. Tenth International Teletraffic Congress– Session 2.4 Paper 2, vol. 1, Jun. 9–15, 1983, Montreal(CA), pp. 1–4, XP002021531 Lewis et al: "Measurements of Repeat Call Attempts in the Intercontinental Telephone Service" cited paragraph 3.

Bell System Technical Journal,vol. 59, No. 3, Mar. 1980, New York US, pp. 295–311, XP000560635 K.S.LIU: Direct Distance Dialing: Call Completion and Customer Retrial Behavior see page 297, left–hand column, line 17– line 38.

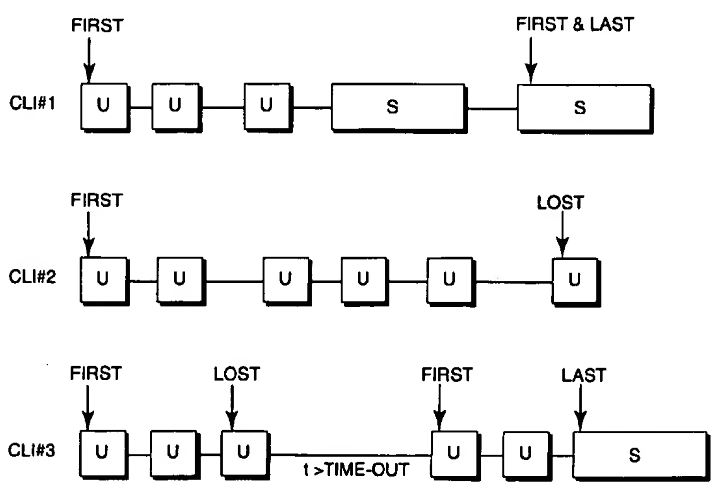
Tenth International Teletraffic Congress–Session 2.2 Paper 7, vol. 1, Jun. 9–15, 1983, Montreal (CA), pp. 1–6, XP002021532 Becker Simcha et al: "Killer Routes and Killer Numbers in Telephone Networks".

Primary Examiner—Curtis A. Kuntz
Assistant Examiner—Duc Nguyen
Attorney, Agent, or Firm—Nixon & Vanderhye P.C.

[57] **ABSTRACT**

In a communications network, call records are generated by a plurality of originators (CLI#1, CLI#2, CLI#3) making calls to a service provider (180). The call records are stored in a database (145) which forms part of the billing function (140) of the network. The call records for the service provider (180) are collated and sent to the service provider for processing. The service provider is provided with a system for analyzing the call records to establish the number of lost callers, rather than the number of lost calls. This is possible since the call records include the CLI information of the originators. This information, gathered over a predetermined period of time, allows the service provider (180) to estimate the number of answering stations (170) necessary to optimize call answering and minimize the number of lost callers.

3 Claims, 7 Drawing Sheets



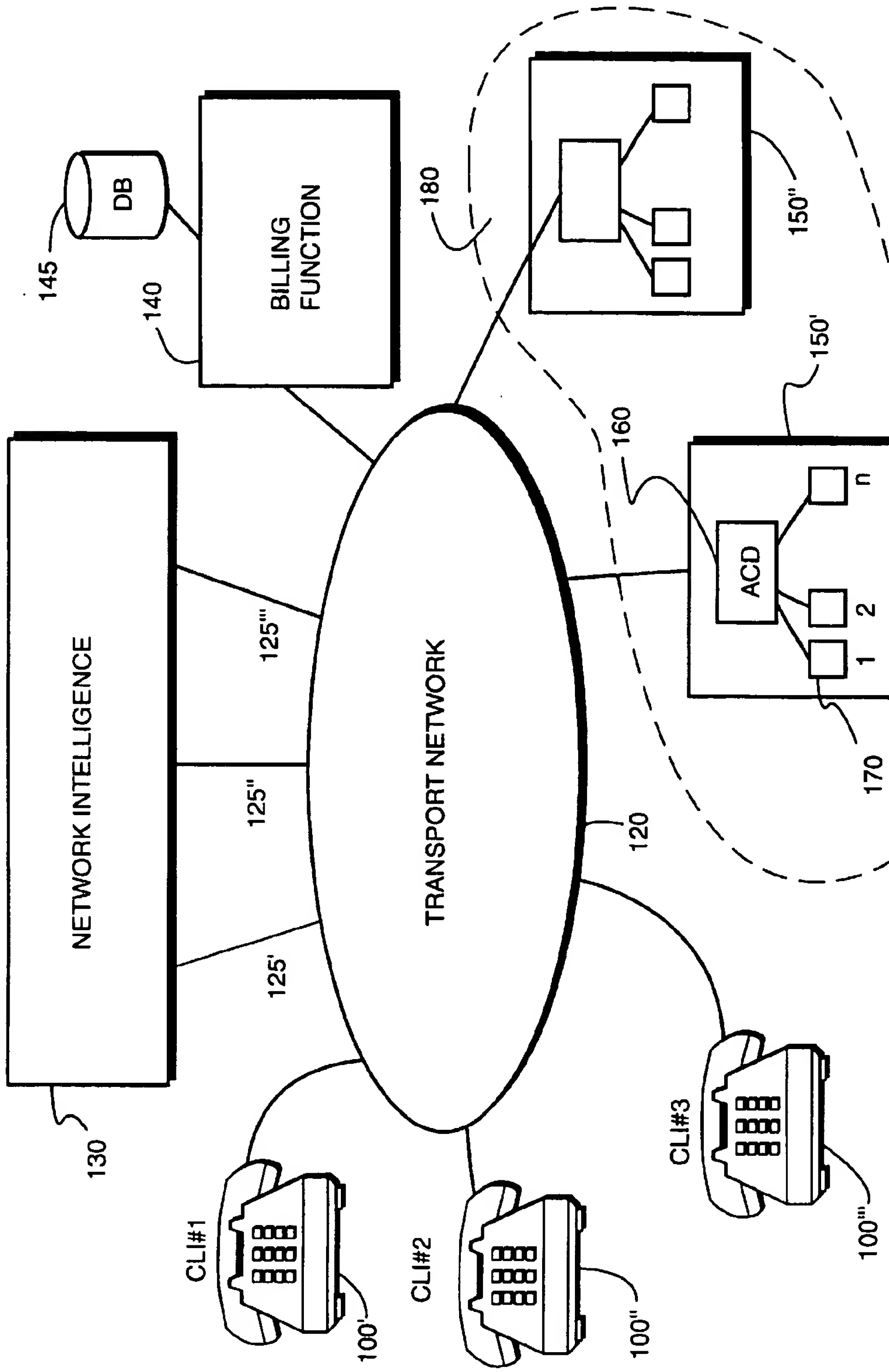

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Fig.1

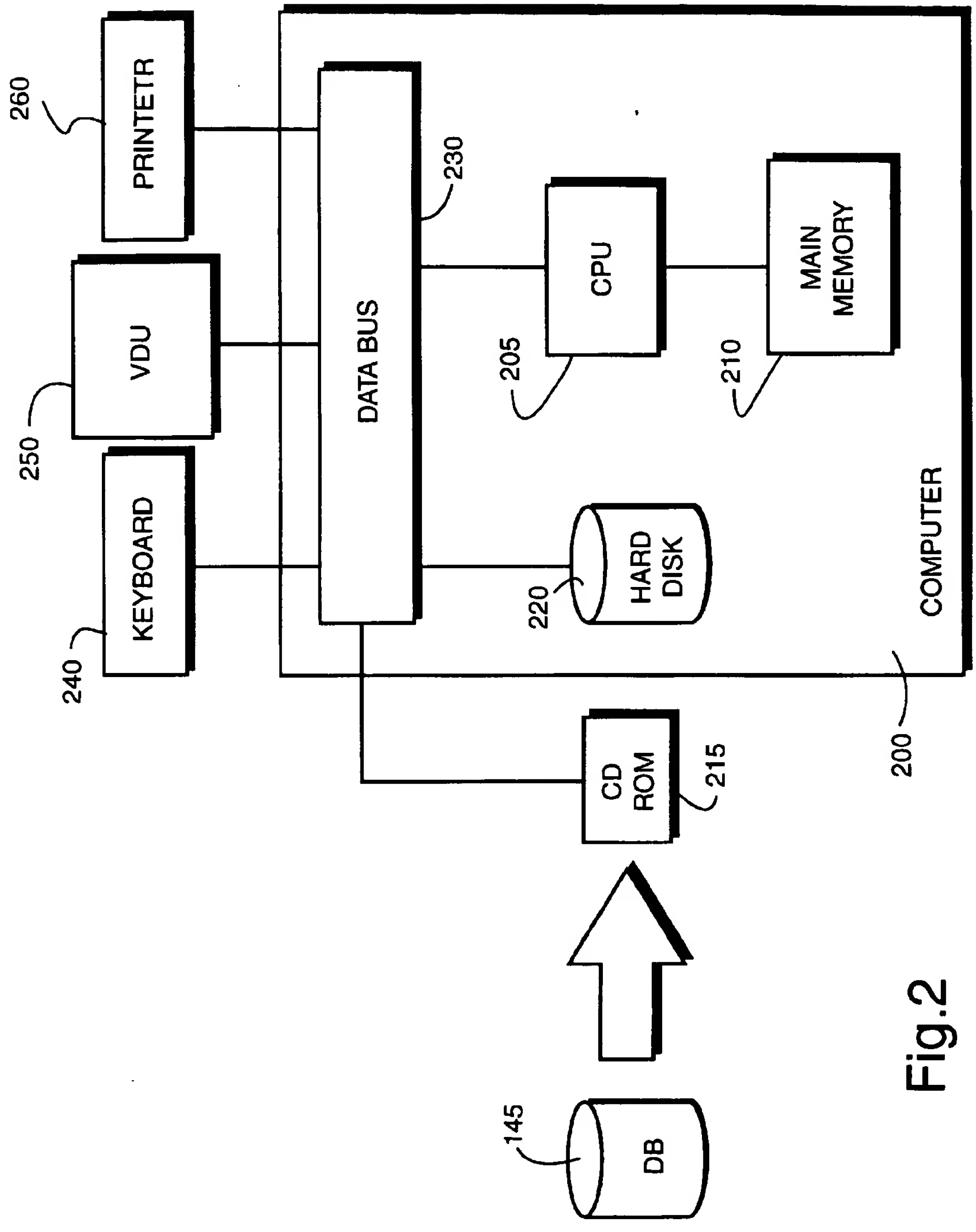

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Fig.2

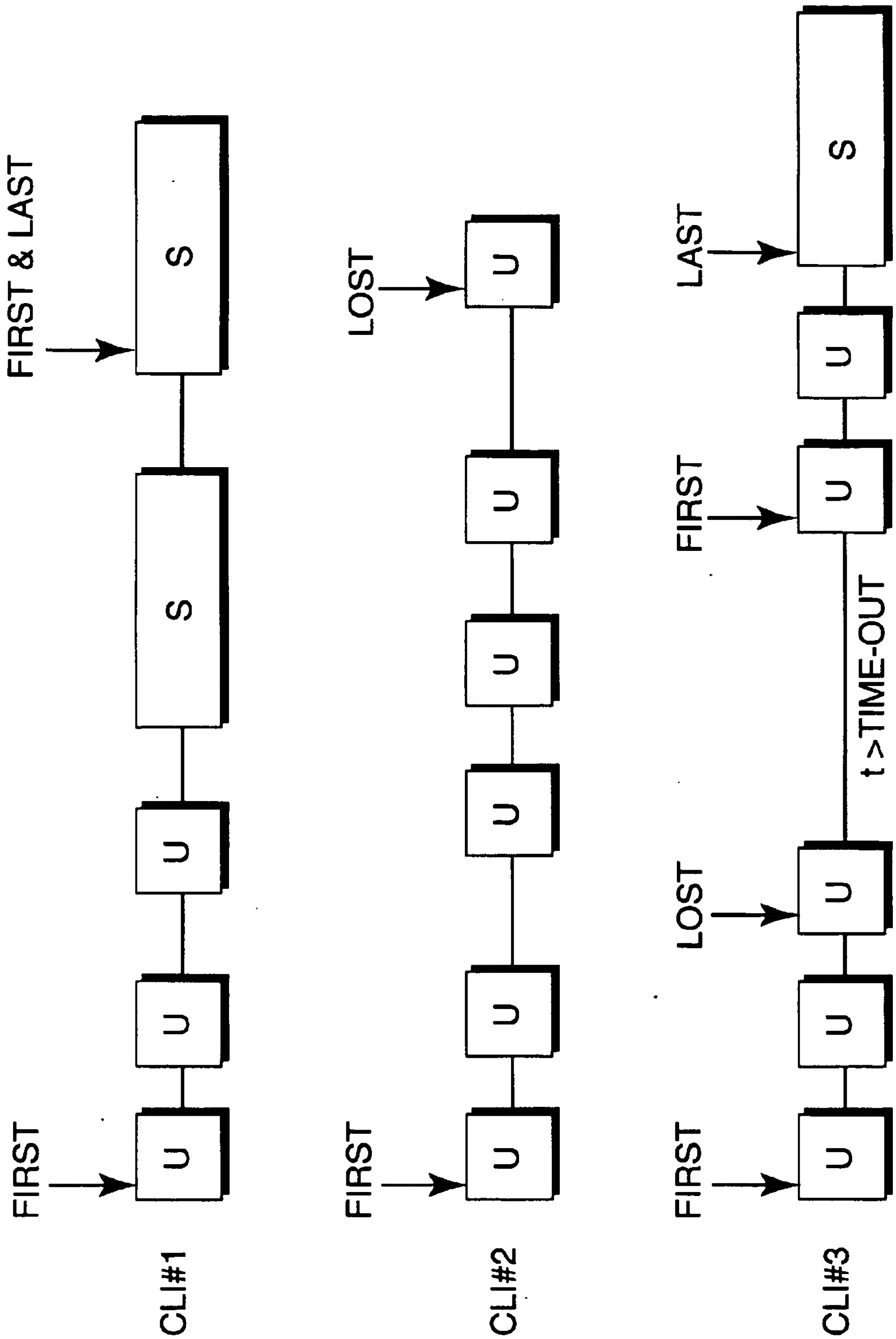
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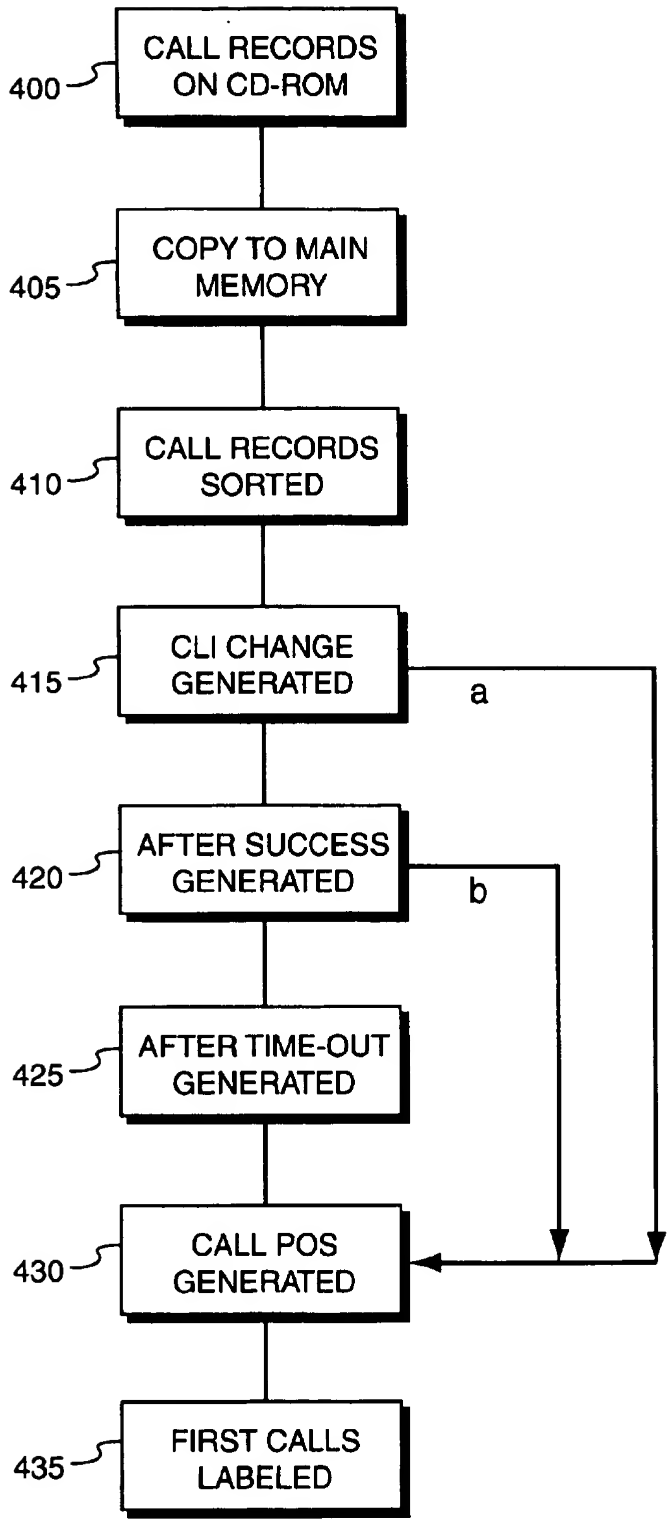
Fig.3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CALL POS GENERATED
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FIRST CALLS LABELED
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Fig.4

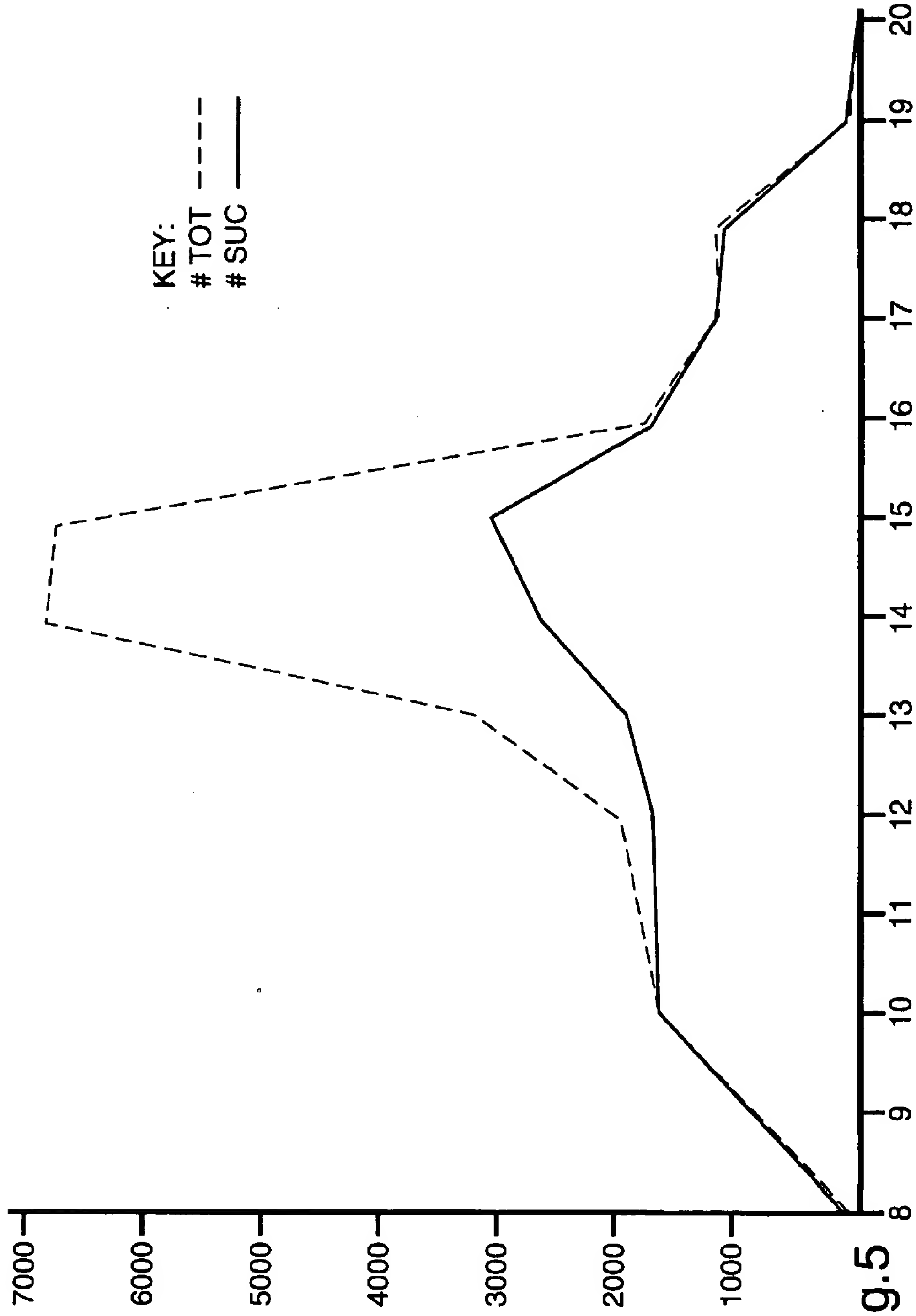
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Fig.5

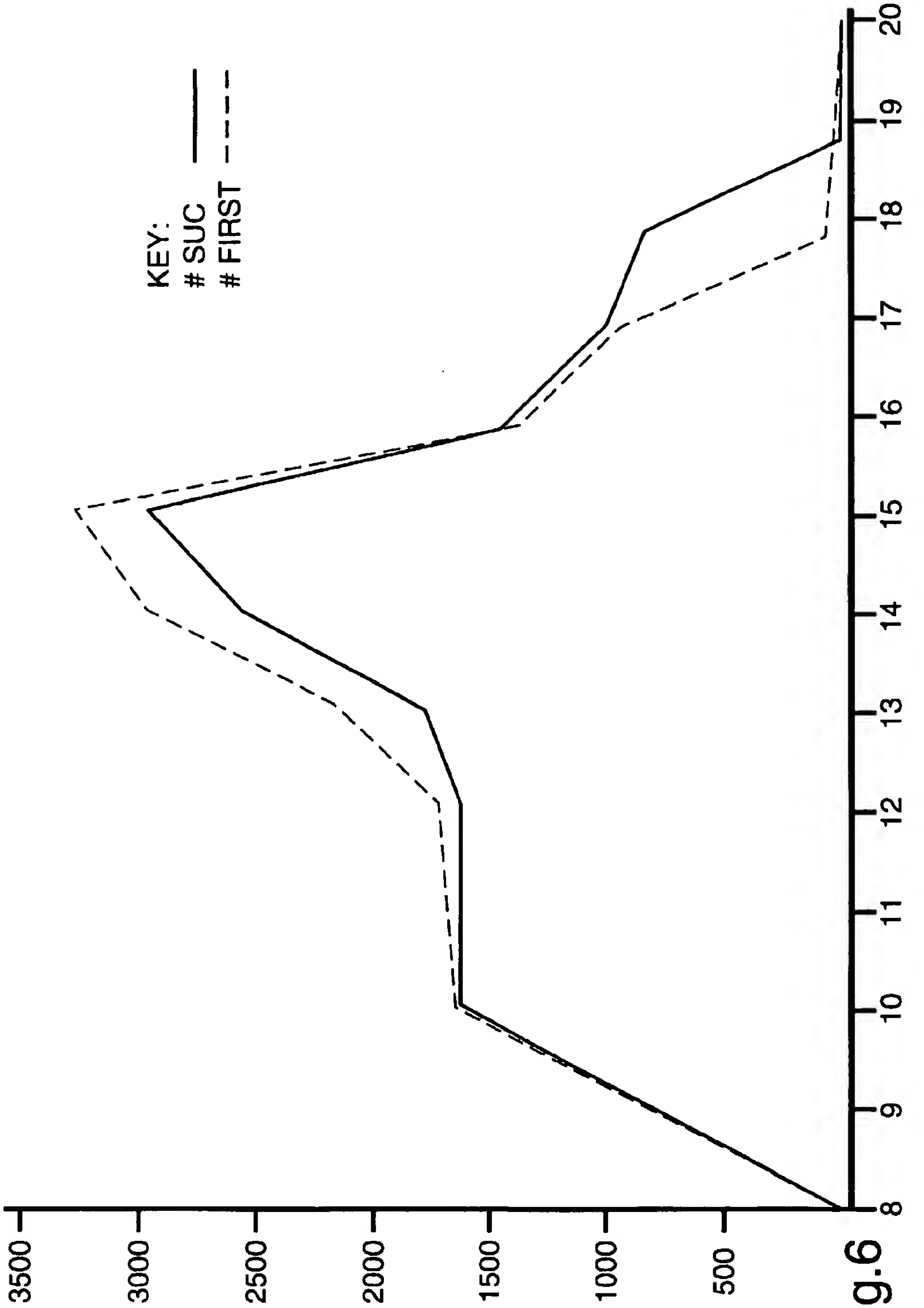
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Fig.6

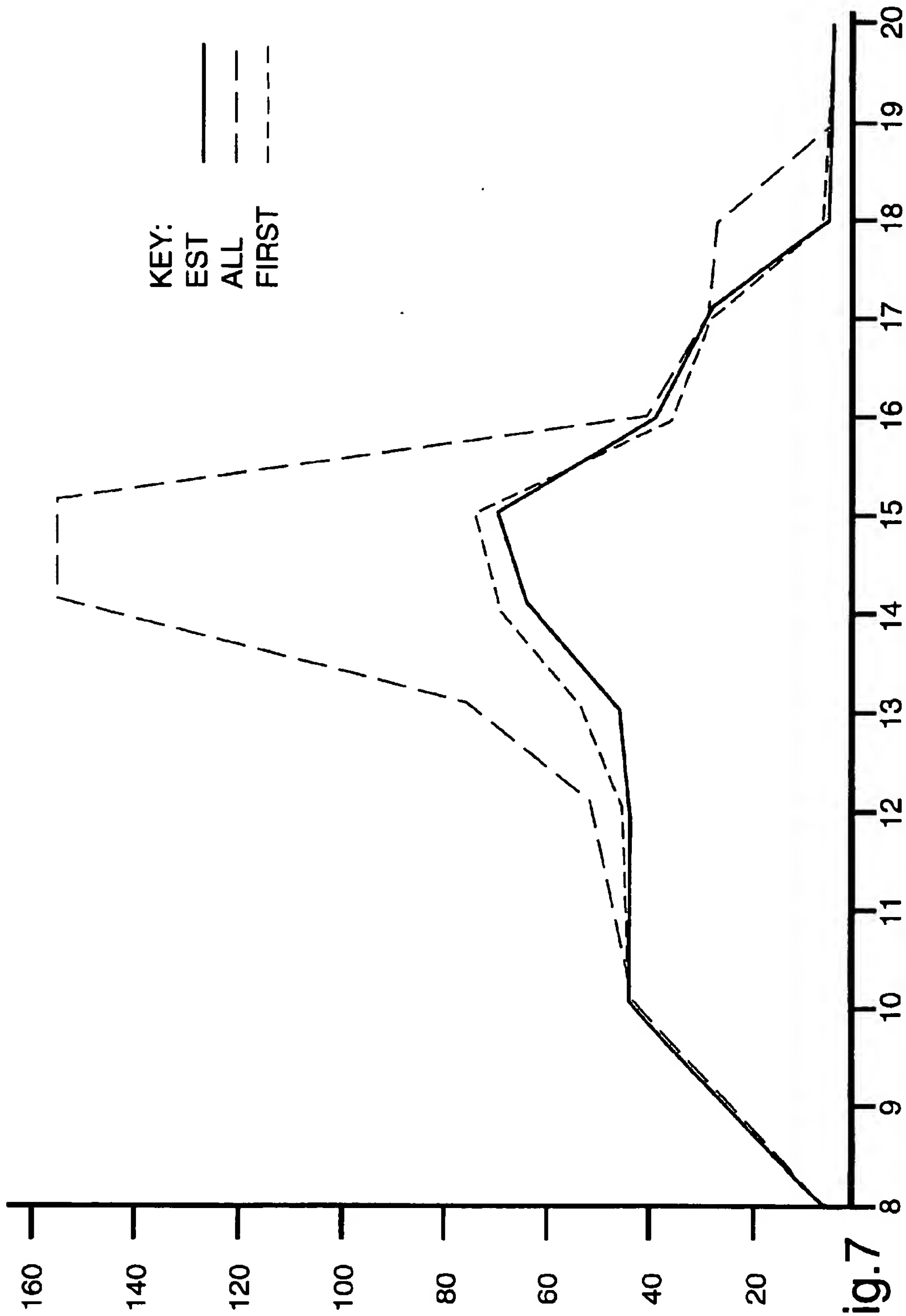
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Fig.7

CALL PATTERNS IN A COMMUNICATIONS NETWORK

The present invention relates to call patterns in communications networks and particularly, but not exclusively, to methods and apparatus for determining call patterns in a communications network for particular call destinations.

Service providers who provide or sell a service by telephone over a communications network, for example insurance service providers, telephone information hotlines or premium rate call services, typically operate answering centres having a number of answering stations for answering calls directed to the service provider. Answering stations may comprise, for example, telephones having human operators for answering calls, or computer-based systems answering calls directly. Computer-based systems most commonly simply play pre-recorded information, however, in some cases, they do provide the facility to interact with a caller via, for example, codes which may be input by a caller using a telephone keypad or via speech recognition.

For service providers whose business relies heavily on telephone custom, it is important to optimise the number of answering stations in operation: too many answering stations are a waste of resources and too few answering stations can result in loss of calls and hence loss of business revenue.

To some extent, the number of answering stations required for an incoming call centre can be forecast by service providers using historic call record data made available by communications network operators. The type of historic call record data available varies, but typically might comprise an indication of the number of successful (effective) calls and the number of failed (ineffective) calls. From this type of call information, it is possible for a service provider to estimate future answering station requirements on the basis of how many calls are being lost.

Call record analysis is known, per se, for measuring call traffic handling performance in telecommunications networks. In "Redialling: a study of subscriber behaviour", C Noordegraph, Het PTT-Bedrijf, vol. 24, no. 1, December 1986 The Hague, historic call data, originating from 96 people from each of 24 designated exchange areas, was collected over a period of four weeks. An analysis of the call record data on the basis of called "destination" in terms of the terminating one hundred group, (because of suppression of the last two of the dialled digits in the process of making the call record information available for the analysis), was used to determine caller behaviour, and in particular the probability that a caller is likely to re-dial in the event of the first call being ineffective. The results lead to the conclusion that increased capacity for call traffic in the network would lead to increased revenue, given that some re-dials were due to congestion in the network and that not all callers were persistent enough to re-dial. The analysis is concerned solely with re-dialling behaviour and such matters as reply probabilities, and is not concerned with issues of traffic volumes.

In "Measurements of repeat call attempts in the international telephone service", by A. Lewis and G. Leonard, Proceedings of the tenth international teletraffic congress—session 2.4 paper 2, vol. 1, Jun. 9–15, 1983 Montreal, historic call record data was analysed for international calls routed through an international gateway. The data was, again, used to determine caller behaviour in the event a first call was unsuccessful.

In accordance with a first aspect, the present invention provides a method of indicating, for a call centre identified by a particular network number and connected to a com-

munications network having a billing function including the generation and storage of call records, the shortfall in the number of answering stations at the call centre required to be active to meet a specified probability that a sequence, as herein defined, will be lost, the method comprising the steps of:

accessing the billing function in accordance with said particular network number and making a copy of respective call records relating to call attempts delivered to said particular network number, each said copy comprising calling number, outcome, and timing information from which the start time and the duration of the call can be ascertained;

sorting said copies into sets such that each set relates to a respective calling number and, where a set contains a plurality of said copies, these copies are ordered chronologically;

for each of said sets which contains a plurality of said copies, dividing the respective ordered copies into sequences of one or more copies such that

(i) if the time interval between an adjacent pair of call attempts of the set is greater than a predetermined value, then the call attempt having the earlier start time is the last of its sequence and the call attempt having the later start time is the first of its sequence, and

(ii) each successful call attempt is the last of its sequence,

designating a sequence as either (a) an answered sequence if it contains a successful call attempt, whether or not that successful call attempt is preceded by one or more unsuccessful call attempts, or (b) a lost sequence if it does not contain a successful call attempt;

counting the answered sequences and the lost sequences;

using the values of call duration and the counts of answered and lost sequences to obtain

a value for the offered traffic in Erlangs,

an actual value for the ratio of lost sequences to total sequences,

and, via the appropriate Erlang formula, an estimate of the effective number of active answering stations;

using said value for the offered traffic, and said specified probability to obtain, via said Erlang formula, an estimate of the number of answering stations which would have been needed to be active to achieve said specified probability that a sequence will be lost; and,

in the event that this latter estimate is greater than the former estimate, providing an indication of the shortfall.

The invention is particularly appropriate for near real time monitoring of the level of service provided by the call centre to generate the indication of the shortfall in the event of, say, an unexpected increase in incoming calls to a call centre of a share dealing company.

Therefore, preferably, there is included the step of transferring the copies from the billing function of the network to an operator of the call centre, and said steps of sorting, dividing, designating, counting, processing, and providing are performed by that operator.

More preferably, the accessing step is such that said transferring step is performed substantially immediately after the billing function has generated the call records.

The method of the present invention could be performed using call records from any network providing the appropriate call record data.

The present invention uses call record analysis for a significantly different purpose than those of the prior art documents described above. The prior art documents describe methods of analysing network traffic, originating

from many sources and directed to many sources, to aid network design. The present invention is directed to optimising call centre answering station numbers. The term "call centre" in the context of the present invention encompasses public service providers, such as Internet Service Providers, and private operators of call centres, such as telephone betting agencies.

The present invention is also significantly different in terms of the call record information required. In particular, the present invention processes only call records directed to a single destination. Also, the present invention relies heavily on the call records containing accurate call origin information. In contrast, neither prior art document is concerned with the volume of traffic offered to a particular destination, and neither describes a system which processes call duration information in combination with a re-dialling analysis, and applies an Erlang formula to provide the shortfall indication as does the present invention.

The present invention will now be described in more detail, by way of example only, with reference to the drawings, of which:

FIG. 1 is a diagram which represents an example of a network configuration;

FIG. 2 is a diagram which represents a system suitable for carrying out the invention;

FIG. 3 is a diagram which shows example call sequences;

FIG. 4 is a flow diagram showing the steps involved in carrying out the invention;

FIG. 5 is a graph showing the total number of calls and the number of successful calls received by an answering centre over a period of one day;

FIG. 6 is a graph showing the number of first calls out of the total number of calls shown in the graph of FIG. 5 and the number of successful calls; and

FIG. 7 is a graph showing Erlang calculations based on the data in the graphs of FIGS. 5 and 6.

FIG. 1 is a diagram representing a communications network. In the network, there are a number of telephones **100** shown connected to a transport network **120**. The transport network **120** is of known type, for example a public switched telephone network, and comprises local exchanges and trunk exchanges (not shown). Typically, although it is not shown in the diagram, the telephones **100** are connected to local exchanges in the transport network **120**, each local exchange is connected to at least one trunk exchange, also known as a digital main switching unit (DMSU), and the DMSUs are typically fully interconnected with each other. Also, at various points in the transport network, there are connections **125** to network intelligence equipment **130**.

The communications network includes a billing function **140**, responsible for receiving call record information from the transport network **120** and for calculating respective bills. The billing function **140** includes a database **145** for storing the call record information.

In practice, the communications network includes other functions, for example network management and service management functions, which for clarity have not been shown on in FIG. 1, but are assumed to be present.

Answering centres **150** are also connected to the transport network **120**. The answering centres **150** may be connected to either trunk or local exchanges, depending on the call capacity requirements of the answering centres **150** and of the transport network **120**. Each answering centre **150** comprises an automatic call distributor (ACD) **160** and a number n of answering stations **170**, each connected to the ACD **160**. The ACDs **160** allocate incoming calls to answering stations **170**. In this case, the answering stations **170**

comprise telephones manned by human telephone operators (not shown). Some ACDs can also allocate calls to a queuing system (not shown) which can hold several calls in a queue until an answering station becomes available. The answering centres **150** together make up a distributed answering centre **180** which in this example represents a called party.

A distributed answering centre **180** comprises more than one answering centre **150**, whereby calls to a single number can be distributed in dependence upon number translation facilities provided by the network intelligence equipment in the communications network to re-direct calls between call centres. For example, for BT's Directory Enquiries service, answering centres are distributed in several locations, and a telephone call to a single Directory Enquiries number, **192**, might be answered in any of the answering centres, depending on a number translation system in the network intelligence which is able to take into consideration call loads in each answering centre.

In operation, a telephone call to a service provider is made by a caller from a telephone **100'**. The telephone call generates a call request when a caller dials the generic number of the service provider. The call request is processed by the network intelligence equipment **130** as follows. The network intelligence equipment **130** relates the generic telephone number of the service provider to there being two possible answering centres **150'** and **150"** each having a different, specific telephone number to which the call can be directed. The network intelligence equipment **130** also has access to information about calls which are currently in progress at each answering centre **150** and the number of answering stations **170**, or the capacity of a queuing system, in each answering centre. From this information, the network intelligence equipment **130** determines to where the call should be directed to be answered, for example to answering centre **150"**, and translates the generic number of the service provider to the specific number of the selected answering centre **150"**. The number is passed to the transport network which routes the call to answering centre **150"** accordingly.

When all answering stations **170** in the answering centres are busy, and any queuing systems are full, the caller will obtain an engaged tone. The caller will then hang up and decide to try again immediately with the hope of being successful, try again later or give up.

All call attempts, whether effective or not, generate call records which are stored by the billing function **140** on the database **145**. According to the present example, call records include:

date & time of the call

duration of the call;

time to answer the call;

network calling line identity (CLI) of the caller

a flag indicating whether a call was effective or not; and

the identity of the call centre handling the call;

FIG. 2 shows the arrangement of equipment required by a service provider to work the invention. The equipment comprises a general purpose computer **200**, such as an IBM compatible 80386 personal computer, and standard peripheral devices such as a CD-ROM drive **215**, a keyboard **240**, a visual display unit **250** and a printer **260**.

The computer **200** comprises a central processing unit **205** which is connected to main memory **210** and a data bus **230**. The data bus **230** provides data transmission paths between the central processing unit **205** and a secondary storage device such as a hard disk drive **220** and interfaces (not shown) to the peripheral devices **215**, **230**, **240**, **250** and **260**.

The invention is controlled by software, for example written in the C programming language, held in the main memory **210** and which is processed by the central processing unit **205**.

Initially, the call records are stored on the database **145** which forms part of the billing function **140** of the communications network. The call records are stored, as they are made available by the transport network **120**, in chronological order, with the CLI of the originator being stored as one field of each call record.

A copy of the call records directed to a particular service provider, held on the database **145**, is supplied to the service provider when the service provider requires its call record information. The call record information is typically transferred to the service provider on a CD-ROM. The CD-ROM is inserted into the CD-ROM drive **215** where the information can be accessed by the central processing unit **205**.

It should be noted that, in some cases, the processing might be carried out by a party other than the service provider, for example by a data mining company which sells respective information to service providers, or even by the communications networks operator itself which might sell the information to service providers or offer the information as a free service.

The central processing unit **205** runs the software embodying the invention and provides results, as described below. The results can be viewed as numerical data or graphical representations of the data on the visual display unit **250**. In the latter case, further commercially available software can be provided to generate suitable graphical data from the computed numerical data. Finally, the results, in whatever form, can be stored on the hard disk drive **220** and printed on the printing device **260** if required.

For the purposes of the present description, a "sequence" of calls is defined as being sequential calls from the same caller (ie from one CLI). One CLI can generate a number of sequences.

FIG. **3** illustrates example call sequences originating from three different CLIs—CLI#1, CLI#2 and CLI#3. If a sequence contains an effective, or successful call S, the call is defined as the last call in a sequence, unless it is the only call, in which case it can be either the last call or the first call as illustrated for CLI **#1**. If a call is ineffective, or is not successful, then it is represented as an unsuccessful call U. If there are further calls from the same CLI, after a successful call S, as illustrated for CLI **#1**, these calls are defined as members of further sequences.

As illustrated for CLI **#2**, the last call in a sequence, if unsuccessful, is classed as a lost call, which ends a sequence.

As illustrated for CLI **#3**, a sequence can also be terminated by an unsuccessful call U followed by a user-defined time period, or time-out period. The duration of the time-out period is adjusted to suit the type of service, and hence the expected type of call pattern, for a particular service provider. The first call after the time-out period is classed as a first call in a new sequence.

In FIG. **3**, "First" calls are defined as the first calls in a sequence, no matter whether they are successful or unsuccessful. "Lost" calls are defined as unsuccessful calls which are the last in a sequence. Also, when queuing systems are in use, a queued call which is eventually answered is classed as successful and a queued call which remains unanswered, due to the caller hanging-up the call, before an answer station becomes available is classed as unsuccessful.

Call sequences may also be defined by other criteria. For example, if the call duration is only expected to be short, for example twenty seconds, a call sequence may be defined as including a maximum of three calls from one CLI in a period of two minutes. Then, if there are seven unsuccessful calls from one CLI in a two minute period, they would count as three call sequences having three lost calls. Other call sequence definitions will become apparent to the skilled person in dependence on the type of service provider service and the duration over which call records are processed. For example, a call sequence definition for a television-triggered telephone competition, where a caller may make many call attempts to the same number in a very short period of time, would necessarily be very different from a telephone-based insurance sales company, where a caller may only make one or two attempts before waiting later to try again, or indeed before moving onto the next insurance company.

In general, the first, or earliest, call of a CLI will act as a basis for defining first calls, with other first call definitions, or sequence definitions being included as significant depending on the requirements of the service provider.

It will be appreciated that CLI information stored by communications network operators might be confidential, particularly if some CLIs belong to ex-directory, or unlisted, origins. Obviously then, it would not be acceptable to disclose the CLI information to anyone else. To overcome this problem, for the purposes of the present invention, the inventors 'scramble' the CLI data using one-way encryption functions (or functions having no inverse) to hide the identities of the originators of the calls. Thus, the CLI information is replaced by tokens, or references, which bear no apparent relation to the original CLIs, whilst the rest of the call record information remains unaltered. Such encryption techniques are generally known and are beyond the scope of the present description.

Table 1 shows call attempts originating from four different CLIs (3228, 4667, 5149 and 6449) over a period of time between 8 am and 5 pm. Obviously, in a real-life situation, calls would be made from many more originating CLIs but, for the sake of simplicity in this example, only four CLIs are shown. Unsuccessful call attempts are a result of there being no free answering stations in operation at the time of the call, typically because all stations are busy with other calls (which originate from other CLIs which are not shown).

Although call records for only one day are included in Table 1, it is important to remember that call records can be made available for any period of time from when suitable call records are available from a communications network to the present. The periods over which the records are analysed is simply then the choice of the service provider, or other party analysing the call records, depending only on when call records become available or for how long the call records are kept.

FIG. **4** shows the steps of the invention. Initially, FIG. **4** shows, in step **400**, data on the CD-ROM stored substantially in accordance with Table 1. The data fields in Table 1 include the CLI of the originator, the time of the call and a flag which indicates whether or not the call was successful, where a "1" indicates that the call was successful and a "0" indicates that the call was unsuccessful, or a failure.

In step **405**, the data is accessed by the central processing unit **205** and copied to main memory **210** where it can be manipulated by the central processing unit under the control of the software. In step **410**, the call records are sorted, first by CLI reference, and then by time for each CLI reference. This has the effect that calls from the same CLI are grouped together and within the groups the calls are in chronological order.

A illustrated in Table 2, in step **415**, a column labelled "CLI Change" is generated which indicates for each call

record whether it is the first record for a new CLI. Then, in step **420**, a column labelled "After Success" is generated which indicates for each call record whether the record follows on from a successful call by the same caller.

A column labelled "After Time-out" is then generated in step **425** which indicates for each record whether the record follows on from a record from the same caller after a duration greater than the time-out period, which in this case is thirty minutes. After the "After Time-out" column, in step **430**, a column labelled "Call Pos" is generated which holds an attribute which is calculated as the position of a call within a call sequence.

The "Call Pos" column values are generated by stepping through the call records from the top, that is to say the end with the lowest CLI and the earliest times for each CLI, and numbering the call records sequentially as the list is stepped through, starting from 1.

As illustrated in Table 2, the count in the "Call Pos" column is reset to 1 for:

a call with a new CLI;

a call after a successful call; and

a call where the time gap between the call and the previous call in the list exceeds the pre-determined time-out period.

Finally, in a step **435**, a column labelled "First Call" is generated which holds a "1" for all first calls in a sequence, where first calls are those having a "1" in the "Call Pos" column. It can be seen from Table 2 that there are eight first calls.

As a general rule, the first call in a list of calls is classed as a first call.

The alternate paths a and b, from steps **415** and **420** respectively, illustrate that the call positions generated in step **435** may only rely on one or two first call sequence definitions rather than all three. The choice of the number and type of first call sequence definitions is determined by the service provider in dependence on the type of business and corresponding call patterns being generated, as described above.

A similar procedure is carried out as that shown in FIG. 4 to identify the last calls in the call sequences, and hence the lost calls. The respective data generated to indicate lost calls is shown in Table 3. Calls from the same CLI are grouped together and within these groups calls are grouped in reverse chronological order.

The initial call record data in Table 3 is the same as that in Tables 1 and 2. The extra information is added in substantially the same way as for Table 2 except that lost calls are highlighted rather than first calls.

The sorted list in Table 3 is stepped through from the end with the lowest CLI and the latest times for each CLI. The calls are numbered sequentially in the "Call Pos" column as the list is stepped through, starting from 1. The count is reset to 1 for:

a call with a new CLI;

a successful call (note that this is different from the secondary set condition in the identification of first calls above in Table 2); and

a call where the time gap between the call and the previous call in the list originating from the same CLI exceeds a pre-determined time-out period (again, this period is set at 30 mins).

The lost calls are the unsuccessful calls that are labelled with a 1 in the "Call Pos" column. It can be seen that there are three lost calls.

The lost calls in Table 3 represent potentially lost customers, or lost business, rather than the number of overall lost calls.

It will be appreciated that the above steps are described by way of example only to explain the procedure. It will, thus, also be appreciated that the actual steps carried out in software might not follow the same specific procedure to arrive at the same results. For example, several steps might be carried out simultaneously. Therefore, it is intended that any procedure which arrives at substantially the same result comes within the bounds of the present invention.

In practice, at some stage, the actual period over which a service provider wishes to analyse the call record data will be defined. The period would depend on how much data is available, how often it is provided and the period of interest to the service provider. It is expected that call record information will be provided by a communications network operator on a monthly basis, although any other period could be defined. In this example, the period is defined simply by the call records shown. In practice it would typically be the periods of peak call rate which would be of interest to the service provider, since it is at peak times where a significant amount of custom is more likely to be lost if too few answering stations are in operation. However, clearly a correct assessment of the number of required operators at times other than the peak is also of interest to the service provider.

Table 4 shows call record statistics generated for a telephone betting agency which took telephone bets for the English Grand National horse race. The number of telephone answering stations was 75. In the table: "#suc" is the number of successful calls; "#unsuc" is the number of unsuccessful calls; "#tot" is the total number of calls (#suc+#unsuc); "#first" is the number of first calls derived as described above; "dur" is the average call duration in seconds; "est" is an estimate of the number of answer stations in operation calculated using the Erlang B traffic formula; "all" is the number of required answering stations calculated using the Erlang B formula and all (#tot) call requests; and "first" is the number of required answering stations calculated using the Erlang traffic formula and only first (#first) call requests.

The graph in FIG. **5** is generated from the data in Table 4. The graph shows the rate at which calls were being made to the call centre (#tot) and the rate at which calls were handled by the call centre (#suc). These statistics, available from the raw call record data directly, show that nearly 7000 calls were attempted in one hour running up to the start of the race at 3.00 pm, but the call centre could only handle a maximum of around 3000 calls in any one hour. Call records generated after the Grand National (after 3.00 pm) relate to bets placed on subsequent races during the day.

The graph in FIG. **6** shows a comparison of the rate at which calls were handled (#suc) with the rate at which first call attempts were made (#first), calculated using the "first call" analysis described above. It is clear from this graph that the actual difference between stimulated and handled traffic is much less than is suggested by the graph in FIG. **5**.

The graph in FIG. **7** shows the results of the Erlang traffic calculations based on the call pattern numbers of the graphs in FIGS. **5** and **6**. The "est" line shows the calculated number of call stations required to handle #suc calls (where the number of call stations was in fact 75). The "all" line shows that if the total number of calls (#tot) was used to calculate the number of call stations required, the number would be roughly double the number actually being used. Finally, using the calculated number of first calls (#first) the "first" line shows that an increase in the number of lines and answering stations of around only 10% would be sufficient to make sure that no calls were blocked.

Traffic analysis using the Erlang formulae to calculate the number of trunks (or call stations) necessary to maintain a

specified level of service has been widely reported and used to maintain communication network service levels for a number of years. One form of the Erlang B formula is shown below:

$$P = \frac{\frac{A^N}{N!}}{\sum_{x=0}^{N} \frac{A^x}{x!}}$$

where

A=offered traffic (Erlangs)

N=number of answer stations

P=probability that a call will be lost

and where offered traffic A is calculated as:

$$A=\lambda h$$

where

λ=the mean rate of call attempts per unit of time

h=the mean holding time per successful call

A detailed discussion of the formulae is beyond the scope of the present description, but the reader is referred, for example, to the text books referenced below for further information, if required. In particular, reference [3] includes computer programs for carrying out the appropriate calculations.

In general, where incoming calls are lost (ineffective) if no answering station is available, the Erlang B formula is appropriate for the calculations. Where incoming calls are queued (in a finite length queue) until a call station becomes available, the Erlang C formula is appropriate for the calculations.

The equations require values of the average calling rate and the average call duration. Average call duration can be calculated from the call duration values available for each call. Such values may be available from the call records directly or from customer billing records.

Thus, from the graph in FIG. 7, it can be seen that the betting agency (the service provider) would only need to increase answering station capacity by about 10% to optimise revenue from betting, by serving substantially all callers using the fewest possible answering stations. Using raw statistics, which simply show the number of lost calls, the betting agency might erroneously feel it necessary to increase answering station capacity by up to 100% to handle every call made at peak betting times.

It is clear from these calculations that the present invention provides an invaluable tool for any undertaking interested in conducting business by telephone. Uses of the invention other than that described above will become apparent to the skilled person. For example, the present invention may be used to calculate the number of answering stations required for a dial-up "logon" point for services such as Compuserve (TM) or the Internet. In these cases, failure to access the network due to there being too few lines, or access points, would result, in the short term at least, in reduced customer satisfaction rather than direct loss of revenue.

Embodiments of the present invention could be arranged to provide a near real time assessment of the number of required answering stations for some service providers. For example, call records could be processed, as described above, to indicate a shortfall in answering stations over, say, the past five or ten minute period. This assumes that call records for calls to a service provider were, for example, transferred to a database accessible by the service provider as soon as possible after the call records were generated. If processing then highlights a need for more operators, surplus operators (for example, those taking a lunch break or an afternoon tea break) could be re-called to man their previously inactive answering stations to meet an increased demand.

Such embodiments might be useful, for example, in a share dealing scenario where shorter term historical data, for example collected over the preceding five or ten minutes as suggested, could be used to cope more effectively with call surges resulting from, for example, unexpected share market movements. Also longer-term historical data, for example collected over days or weeks, would be used to plan for normal answer station requirement variations, for example, to cope with normal peaks at the beginning and at the end of the share dealing day.

Presently, such rapid call record access is not feasible. However, this does not preclude more rapid access becoming a reality in future and, as such, the present invention is intended to encompass all such eventualities.

References

[1] "Telecommunications Networks", IEE Telecommunications Series 1, edited by J E Flood, first published 1975.

[2] "Communication Switching Systems", Rubin & Haller, Reinhold Publishing Corporation NY, Chapman & Hall Ltd, London, 1966.

[3] "Local Networks", William Stallings, Third Edition, Published by Collier Macmillan Canada, Inc. 1990.

[4] "Telecommunications Engineering", J. Dunlop & D. G. Smith, Second Edition, Published by Van Nostrand Reinhold, 1989.

Tables

TABLE 1

CLI	Time	Success 1 Failure 0
4667	08:59:01	1
5149	10:12:14	0
5149	10:13:01	1
4667	11:31:27	0
4667	11:32:10	0
4667	11:32:56	0
4667	11:33:21	1
6449	12:27:58	0
6449	13:33:46	0
6449	13:35:02	1
5149	14:01:21	0
5149	14:03:17	1
5149	15:27:40	0
3228	16:08:13	0

TABLE 2

CLI	Time	Success 1 Failure 0	CLI Change	After Success	After Time-out	Call Pos	First Calls
3228	16:08:13	0	1			1	1
4667	08:59:01	1	1			1	1
4667	11:31:27	0		1	1	1	1
4667	11:32:10	0				2	
4667	11:32:56	0				3	
4667	11:33:21	1				4	
5149	10:12:14	0	1	1		1	1
5149	10:13:01	1				2	
5149	14:01:21	0		1	1	1	1
5149	14:03:17	1				2	
5149	15:27:40	0		1	1	1	1

TABLE 2-continued

CLI	Time	Success 1 Failure 0	CLI Change	After Success	After Time-out	Call Pos	First Calls
6449	12:27:58	0	1			1	1
6449	13:33:46	0			1	1	1
6449	13:35:02	1				2	

TABLE 3

CLI	Time	Success 1 Failure 0	First Call Count	CLI Change	Success	Time Gap Exceed	Call Pos	Lost Call
3228	16:08:13	0	1	1			1	1
4667	11:33:21	1	4	1	1		1	
4667	11:32:56	0	3				2	
4667	11:32:10	0	2				3	
4667	11:31:27	0	1				4	
4667	08:59:01	1	1		1	1	1	
5149	15:27:40	0	1	1			1	1
5149	14:03:17	1	2		1	1	1	
5149	14:01:21	0	1				2	
5149	10:13:01	1	2		1	1	1	
5149	10:12:14	0	1				2	
6449	13:35:02	1	2	1	1		1	
6449	13:33:46	0	1				2	
6449	12:27:58	0	1			1	1	1

TABLE 4

Time	#suc	#unsuc	#tot	#first	dur	est	all	first
8	60	10	70	65	75	5	6	5
9	807	3	810	785	74	25	25	25
10	1661	19	1680	1639	82	45	45	44
11	1660	135	1795	1687	90	45	48	45
12	1656	354	2010	1723	94	45	52	46
13	1814	1320	3134	2105	86	48	77	55
14	2572	4232	6804	2920	68	65	154	72
15	2949	3714	6663	3191	63	73	154	78
16	1462	82	1544	1369	70	40	42	38
17	1023	9	1032	962	90	30	30	29
18	889	42	931	103	69	6	28	7
19	56	16	72	63	75	5	6	5
20	48	7	55	47	53	5	5	5
21	42	4	46	39	60	4	5	4
22	29	0	29	29	60	4	4	4
23	4	0	4	2	60	2	2	2

We claim:

1. A method of indicating, for a call center identified by a particular network number and connected to a communications network having a billing function including the generation and storage of call records, the shortfall in the number of answering stations at the call center required to be active to meet a specified probability that a sequence, as herein defined, will be lost, the method comprising the steps of:

accessing the billing function in accordance with said particular network number and making copies of respective call records relating to call attempts delivered to said particular network number, each said copies comprising calling number, outcome, and timing information from which the start time and the duration of the call can be ascertained;

sorting said copies into sets such that each set relates to a respective calling number and, where a set contains a plurality of copies, these copies are ordered chronologically;

for each of said sets which contains a plurality of said copies, dividing the respective ordered copies into sequences of one or more copies such that

(i) if the time interval between an adjacent pair of call attempts of the set is greater than a predetermined value, then the call attempt having the earlier start time is the last of its sequence and the call attempt having the later start time is the first of its sequence, and

(ii) each successful call attempt is the last of its sequence, designating a sequence as either (a) an answered sequence if it contains a successful call attempt, whether or not that successful call attempt is preceded by one or more unsuccessful call attempts, or (b) a lost sequence if it does not contain a successful call attempt;

counting the answered sequences and the lost sequences;

processing the values of call duration and the counts of answered and lost sequences to obtain

a value for the offered traffic in Erlangs,

an actual value for the ratio of lost sequences to total sequences, and, via the appropriate Erlang formula,

an estimate of the effective number of active answering stations;

processing said value for the offered traffic, and said specified probability to obtain, via said Erlang formula, an estimate of the number of answering stations which would have been needed to be active to achieve said specified probability that a sequence will be lost; and,

in the event that this latter estimate is greater than the former estimate, providing an indication of the shortfall.

2. The method as claimed in claim 1, including the step of transferring the copies for the billing function of the network to an operator of the call centre, and said steps of sorting, dividing, designating, counting, processing, and providing are performed by that operator.

3. The method as claimed in claim 2, wherein said accessing step is such that said transferring step is performed substantially immediately after the billing function has generated the call records.

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